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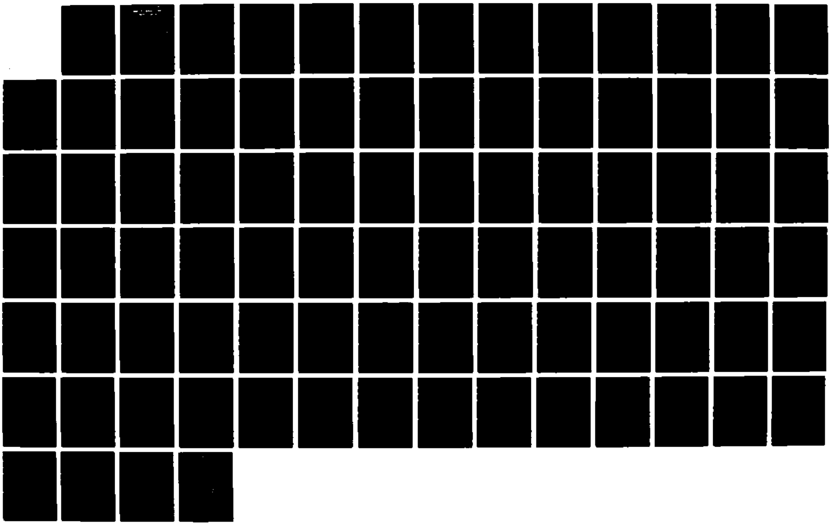

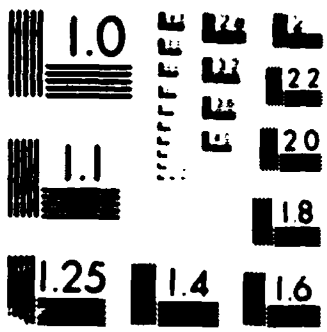

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NAVAL POSTGRADUATE SCHOOL
Monterey, California

THESIS

AN EVALUATION OF INTERACTIVE LABORATORY
SYSTEM SOFTWARE, ILS
PC/DOS, A DIGITAL SIGNAL PROCESSING
SOFTWARE PACKAGE

by

Thomas A. Quintero

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Thesis Advisor D. E. Kirk

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An Evaluation of Interactive Laboratory System Software, ILS PC DOS, a Digital Signal Processing Software Package

by

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I. DIGITAL SIGNAL PROCESSING SOFTWARE

A. INTRODUCTION

The purpose of this study is to evaluate the Interactive Laboratory Systems (ILS-PC) digital signal processing software package. This is accomplished in three ways. First, by identifying the need for digital signal processing software and the basic computational operations that a software package of this type should perform. Secondly, by describing the operation of the software and development of a conceptual thought process that incorporates ILS in solving digital signal processing problems. Finally, by using ILS to solve a number of problems and evaluating the software package by comparing its capabilities and limitations with existing tools available for digital signal processing at the Naval Postgraduate School.

B. NEED FOR DIGITAL SIGNAL PROCESSING SOFTWARE

As modern technology tends more toward digital methods in signal processing and analysis, the need for software that consolidates and decreases the computational burden of these methods becomes apparent. Time savings are realized if the software is interactive and requires only input parameters. From an academic perspective, use of this software can improve instructional effectiveness and increase the research productivity of students. Solutions that require redundant computations such as, for example, determining the frequency response of a filter, can be completed without substantial programing. This increases the flexibility and availability of computer generated solutions for analysis. Finally, use of this software helps the student remain abreast of available analysis tools.

C. DIGITAL SIGNAL PROCESSING SOFTWARE PACKAGE REQUIREMENTS

1. Personal Computer Based Software

The first requirement is for the software to be compatible with personal computers. While the obvious trade off between this and a mainframe installation is a loss in computational precision and available memory, there are a number of advantages. The most important advantage is the flexibility and portability of personal computers. First, the user can purchase the software commercially. This eliminates dependence upon software packages only available on the mainframe system. Using

personal computers eliminates the user's need to transfer programs between different computer systems and the compatibility adjustments to the programs that may be encountered in doing this. Personal computers can also be upgraded as necessary to meet the individual needs of the user. This presents the potential for the creation of a tailored library of signal processing programs.

Of equal importance is availability and cost savings when compared to a mainframe system. A personal computer based system is subject to the user's schedule. Unlike a mainframe system, there is no competition for CPU time or unavailability due to scheduled maintenance and unscheduled down times. Also, there is no need to buy computer time.

2. Signal Processing Capabilities

The requirements for a signal processing software package depend on its intended use, however, there are some basic capabilities that will be needed for all situations. The software should be flexible enough to allow the user to obtain the solution of most signal processing problems. The following requirements are intended to be a basis of what is expected of a signal processing software package:

- Interactive Interface. The software should allow the user to describe signals or systems through the use of a keyboard or other interactive interface.
- Signal Generation. There should be the capability to create or form combinations of common sequences such as sinusoids, exponentials, samples, or other similar sequences.
- External Data Input. The software should accept input data from an external source or disk file for analysis.
- Graphical Output. The software should provide well-labeled graphical output of signals or analysis results on both an interactive video screen or in hard copy at the user's option.
- Extendability. The software should allow the user to describe new operations and add these to the system.
- Time Domain Operations. The software should be able to perform common time domain operations, such as convolution and filtering.
- Frequency Domain Operations. The software should be able to perform common frequency domain operations, such as radix 2 FFT and frequency responses.
- Random Signals. The software should be able to generate random signals with user-specified characteristics.
- Determination of Statistical Characteristics. The software should be able to compute statistical characteristics of signals, such as probability density functions, correlation functions, cross correlation, etc.

- Processing of Random Signals. The software should be able to process random signals through linear systems and compute random signal outputs and their statistical characteristics.

The requirements listed are representative of most of the tools required for effective signal processing. The list is not intended to determine the exact needs of the software but rather act as a guide to selecting optimum software. [Ref. 1]

II. USING ILS

Having established the requirements for digital signal processing software, the analysis of a software package can begin. Interactive Laboratory System (ILS-PC) was purchased by the ECE department for this purpose. The scope of this analysis is to describe the operation of ILS and from this develop a general process to facilitate using the software. Initially, this evaluation of ILS was conducted using ILS version 5.0, however, a newer version, ILS version 6.0, was acquired late in the development of this study. To incorporate the improvements and changes of this new version, the version 6.0 capabilities are listed along with a discussion of the newer ILS menu mode of operation.

A. DESCRIPTION OF ILS

ILS is a personal computer based digital signal processing tool. The software can operate in two modes, the command line mode and the menu mode. The signal processing functions that ILS performs can be summarized in eight areas. Discussion of these topics reinforces the utility and capability of ILS as an independent signal processing work station.

1. Operation of ILS

As previously mentioned, ILS is completely interactive. The software identifies commands using three alphabetic characters as mnemonics. These commands can be viewed as subject identifiers for the signal processing applications they perform. The commands are alphabetically listed in the ILS Command Reference Guide which also provides information on their specific usage. This information consists of the command's function, format, input requirements, generated output, and arguments. The arguments, which are alphabetic and numeric, determine the specific operation the command is to perform. Several examples are also provided to demonstrate some applications of the command. Appendix A contains an excerpt from the ILS Command Reference Guide for the design filter command, DFI. This command designs a Butterworth, Chebyshev, or elliptic filter. In this command, the alphabetic arguments allow the design of band pass filters for octave filtering, or the storage of the designed filter in either the COMMON file or as a record in a primary or secondary file or both. The numeric arguments determine the type of filter to design based on the

user's specification of sampling frequency, stopband attenuation, passband attenuation, band edges and the order of the filter. The output consists of very useful information such as the pass band and stop band edges, the sampling frequency, the transfer function of the filter, the poles and zeroes, and the first and second order quadratic factors of the transfer function. [Ref. 2: p. 6-5]

The utility of this command is extensive, considering the many different types of filters that can be designed; however, it is only a small part of the signal processing analysis capability of ILS. Complete signal processing analysis with ILS requires the successive use of different commands. For example, the frequency response of a filter transfer function generated by the DFI command and stored in a record file can be determined by using the Fast Fourier Transform (FFT) command. Successive commands do not have to be related to the design process. Examples of this are the List Records (LRE) command, which lists the contents stored in a record file and the Display Records (DRE) command which graphically displays a record file on the terminal screen.

a. Command Line Mode

The command line mode operation of ILS is meant for the experienced user. Use of this mode requires the user to manipulate the ILS commands and their arguments to perform the desired analysis. There are two ways to operate ILS in this mode. The first method is to execute successive commands, applicable to the required analysis, one command at a time, examining the intermediate results obtained by performing each command. This gives the user an opportunity to inspect each part of the analysis process, preventing the cascading of errors. The second method is operation of the command line mode with a delimiter. This allows the user to enter a number of commands on the command line to be performed as one command. This gives the user an ability to perform repetitive analysis of a system much more efficiently than with the first method. Since there are approximatly 90 ILS commands to choose from, operation in this mode can become both tedious and time consuming. A method which can simplify the selection of appropriate ILS commands for analysis is the menu mode. [Ref. 2: pp. 6-1 and 7-7]

b. Menu Mode

The ILS menu mode is designed to help beginners to use ILS. There are many menus that make up this mode. The menu structure is treelike, allowing the user to make choices which narrow general categories until eventually data entry for a

particular signal processing application is performed. Each menu item has a function key associated with it for selection. If ILS command argument entry is required, the screen displays the arguments for data entry.

The menu mode of ILS operation has several advantages. First, there is no need to open or create file space for the intermediate results of commands used in the signal processing analysis; this is performed automatically. The menus narrow the selection of ILS commands to choose from in terms of the desired signal processing application. The obvious disadvantage of working with a menu driven system is having to prompt through each of the menus for processes which require the redundant selection of commands. Because of this, ILS allows the user to jump directly to a specific menu using the direct access facility or between the menu mode and the command line mode. This gives the user access to the advantages of both ILS subsystems. [Ref. 2: p. 4-5]

2. Capabilities of ILS

The capabilities of ILS can be organized into eight categories. The following is a list these categories and some of the functions available through the software.

1. Data input output. ILS allows the user to acquire and playback data through A D and D A hardware, generate waveforms such as sinusoids, exponentials, noise and speech, and convert data from external files in ASCII or binary format into ILS format for analysis and reconvert to its original form for transfer back.
2. Waveform Display. ILS can display data with a standard XY or three dimensional plot. There are also options for overlay plotting and expanding of segments of data for display.
3. Numeric Listing. ILS allows the user to list the numeric contents of datasets or ILS files.
4. Data Manipulation and Editing. ILS allows the user to convert data between coordinate systems, such as polar to rectangular, scale data through multiplication or addition of an offset, modify particular points in a file, or shift data to represent a delay.
5. Frequency Analysis. ILS allows the user to perform functions such as computing a Fourier transform using a radix-2 FFT, coherence analysis to generate normalized cross spectra, or transfer function estimation, which allows the user to estimate a transfer function and review its Bode and Nyquist plot.
6. Digital Filtering. ILS allows the user to design recursive and non-recursive filters, compute and graphically display their frequency responses, and allows filtering of data through the designed filter for analysis.

7. Numerical Analysis. ILS allows the user to average, integrate or differentiate data. Other applications include performing convolution, correlation and waveform arithmetic.

8. Speech Processing. ILS allows the user to perform speech processing analysis such as linear predictive coding analysis or pitch analysis. There are also options to display and review speech data, spectra or vocal tracts. [Ref. 2: pp. 4-14 to 4-19]

3. Summary.

A general description of the operation and capabilities of ILS has been presented. The signal processing commands in this package consists of a number of self-contained programs. This is advantageous since it allows the user to execute a series of these commands in performing signal processing analysis. Minimal programming is required, since the ILS commands only require the input of arguments which represent the problem parameters. The ILS commands are also general and flexible enough to be tailored to almost any signal processing application.

B. GUIDE TO PROBLEM SOLVING WITH ILS

As with any application of a computer software package, a systematic approach can be developed to facilitate the use of ILS. This process is best understood after a review of general problem solving techniques.

1. Problem Solving Technique

The first requirement in solving any problem is to identify certain parameters. It is essential to identify what is given, what needs to be determined, what are the relationships pertinent to the problem, and what assumptions are to be made. Having answered these questions, an ordered sequence of steps can be followed which leads to the solution of the problem. This can involve many things such as simplifications of equations or programming of redundant operations. The final step in the process is to examine the results and determine their validity and acceptability as a solution.

2. Problem Solving Techniques With ILS

Determining solutions with ILS follows the same process, however, ILS limits the analysis techniques available for signal processing. This requires the user to manipulate ILS software to get a solution. The problem solving process with ILS is summarized as follows:

1. Determine what is given or available in the problem statement.
2. Identify from the problem statement what must be determined.

3. Identify what ILS commands would be useful in solution of the problem. This is more easily accomplished using the ILS menu mode.
4. Order the ILS commands required in solution of the problem.
5. Execute the commands of part (4) and examine the results. Repeat steps 3-5 as necessary until the solution is satisfactory.

This process is short due to the interactive structure of ILS. It also allows the user to experiment and determine an optimal solution.

C. PROBLEM SOLVING WITH ILS

1. ILS Files

With ILS, the use of successive commands in the analysis process produces a requirement for storage of intermediate results in files. ILS works through the manipulation of these files. ILS files are also used to store data input from an external circuit and data transfered from other computer files. The type of data stored in a file determines the type of file it is and there are five different types of data. File manipulation and generation are the basis for successfully using ILS.

a. COMMON File

The COMMON file is required at all operating levels of ILS, storing information to interface and operate the software such as parameter definitions, constants, and status flags. Changes to the contents of the COMMON file are by default written over the previous contents allowing for automatic updating of system parameters. Since this file is always read first when booting ILS, the user always starts exactly where the previous session ended. The COMMON file also has the utility of being a scratch pad for storage of intermediate results. Data from the COMMON file can be extracted and manipulated by transfering it to an ILS applicable type of file using appropriate ILS commands. [Ref. 2: p. 6-2]

b. Other Files

The following is a list of other files internal to ILS:

1. Sampled Data Files. Data for these files is always in integer format and is created using any of three ILS data sources:
   - Test Signals and Functions. ILS has commands that need output files to store their results. These commands generate sampled data files by sampling ILS generated waveforms created by the user.
   - External Files. ILS can be used to create data files from external files in ASCII format, coded ASCII format, or binary format. This allows the user to input data that can be analyzed using ILS commands. There are requirements for the format of the external data. The ILS Command Reference Guide contains the pertinent details of these conditions.

- Analog Waveform. ILS has the capability of accepting an analog source as input to create a data file. With ILS compatible A D and D A hardware, the software can convert external signals to ILS format for analysis. This is not explored by this thesis. [Ref. 2: p. 8-8]

2. Analysis Files. Analysis files store data created by executing speech analysis programs on segments of sampled data. The data is integer data in vector form.
3. Record Files. Record files can store two types of data. The first type is signal processing data. These data consist of real or complex points and is representative of a time or Fourier series. The second type is feature data, which is a matrix or vector representation of properties of experimental data.
4. Label Files. Label files store label data which is ASCII data describing in code and words the location of significant events or segments in sampled data files. [Ref. 2: p. 6-3]

2. Using ILS Files

Binary files are the files that ILS uses the most. These files are the sampled data files, record files and analysis files. All of these files are structured the same. The lengths of the ILS files are set by the user and only limited by the available memory on the disk used. The length is expressed in disk blocks, 512 bytes per block. The file blocks are numbered from 0 to N-1, where N is the number of blocks in a file. The ILS command, FIL, is used to create, delete, select, or unprotect a file. The details of the use of this command are contained in the ILS Command Reference Guide and are demonstrated in the next chapter. An important point in using ILS files is how they are selected and identified. ILS files are identified by two alphabetic characters and a file number (1 - 9998). The two letter prefix is used to represent a single file or a group of files. Each ILS file may be selected with one of six different pointers. These pointers are primary A, B, or C, and secondary A, B, or C. Which pointers to use depends on how the file is used by ILS commands. This is best explained by the following example in which two time series record data files are convolved using the Convolution (CNV) command. According to the command requirements, one file must be declared a Primary (A) file and the other a Primary (B) file; however, a Secondary (A) file must also be opened in order to store the results. In this case the FIL command must be used three times, first to declare one file as primary (A), second to declare the other file as Primary (B), and finally to declare a file as secondary (A) to store the results. This allows the user to save the original data for further analysis, if desired. For instance, the file declared the Primary (B) file could be redesignated the Primary (C) file and the contents altered. The file can then be redesignated as the

Primary-B file and the convolution performed again. It makes the files easy to use, however, keeping track of the files and what each contains may become a problem. More detailed examples to help understand the use of ILS files will be presented in the following chapter. [Ref. 2: pp. 8-3 to 8-5]

III. SOLVING PROBLEMS WITH ILS

The purpose of the following examples is to demonstrate the use of ILS in digital signal processing. The problems show in detail the terminal input and output of running the ILS commands in the command line mode and outline the use of the problem solving techniques developed in Chapter II. The format that ILS commands require is outlined in Chapter 6 of reference (2). The meanings of the command arguments are outlined in the ILS Command Reference Guide.

A. DIGITAL FILTER DESIGN

The following three examples display different methods of using ILS to design recursive and non-recursive digital filters.

Problem 1: Using the bilinear transform procedure and a sampling frequency of 50K Hz, determine a minimum order, bandpass Butterworth digital filter that meets the specifications shown in figure 3.1 and plot the frequency response [Ref. 3: p. 714].

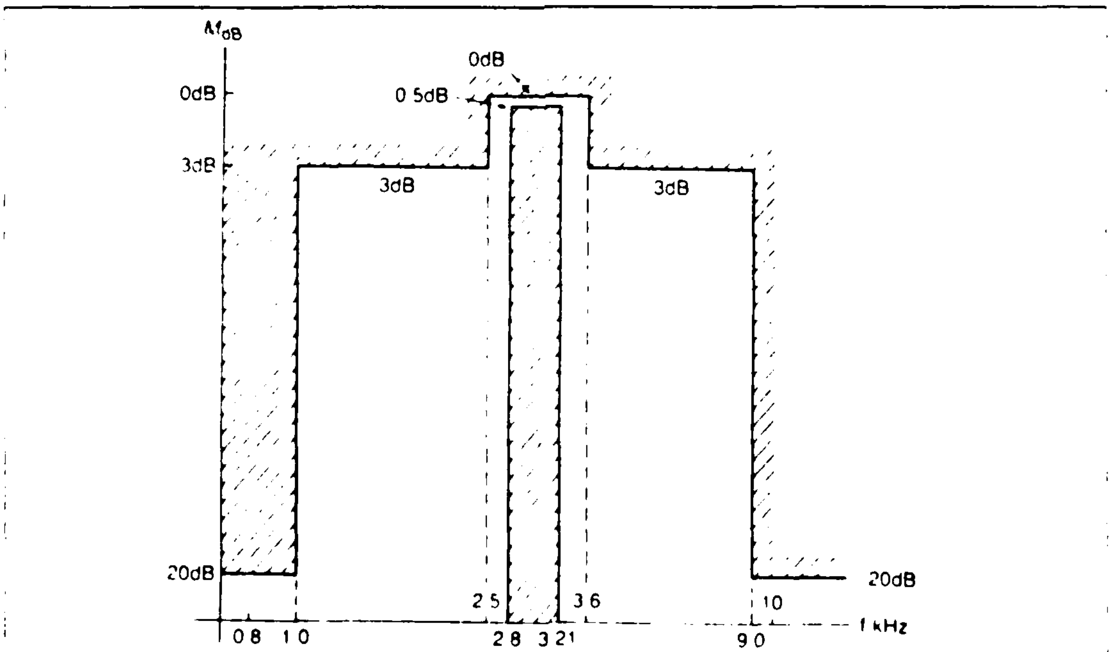

Figure 3.1 Design Specifications Problem 1 [Ref 2: p. 714].

The analysis begins with determining what is given in the problem statement that can help in designing the filter. In this problem, the given information includes the sampling frequency of the filter and the stopband, cutoff and passband attenuations and frequencies for the filter. The next step is to determine what must be found to plot the frequency response. This requires determining the order of the filter and the transfer function. The third step is to determine what ILS commands are to be used in the the solution of the problem. The following ILS commands apply:

- File Command (FIL). This command is used to list, select, create, unprotect, or delete a file.
- Open Record File Command (OPN). This command is used to create and initialize primary and secondary files.
- Design Filter Command (DFI). This command is used to design an elliptic, Butterworth, or Chebyshev filter.
- List Records Command (LRE). This command lists signal processing records from consecutive primary or secondary files.
- Fast Fourier Transform Command (FFT). This command is used to perform Fast Fourier Transform operations on records.
- Display Record Command (DRE). This command is used to display signal processing record files.

In this problem, the FIL and OPN commands must be used first to create storage for the record data that will be generated using the DFI and FFT commands. Next, the DFI command is used to determine the transfer function of the filter. The FFT command is then used to determine the frequency response and DRE displays the results. The LRE command can be used any time during the analysis to list the record data generated by DFI or FFT. The order in which the ILS commands are used is as follows:

1. Initialize two files to store the results of the DFI and FFT commands. This is done using the FIL command with the DEY argument which deletes any data in the file TQ100. The first numeric argument is the numerical filename of the file and the second is the number of consecutive files to delete starting with TQ100. ILS will by default make file TQ100 the primary (A) file.

   Input: *FIL DEYTQ100,,2*

   ```
   ILS responds with: TQ100.          DOES NOT EXIST
                      TQ101.          DOES NOT EXIST

                      TQ100.          DOES NOT EXIST
                      PRIMARY FILE
   ```

2. The files are prepared to accept data. The DFI command can store the transfer function of the filter as a record in the COMMON file or a primary or secondary file. If the transfer function of the filter is stored as a record in the COMMON file, only the Frequency Display (FDI) command can be used to compute and display the frequency response. This command automatically scales the display but does not produce a numerical listing of the frequency response thereby limiting the user to relying on the display to interpret how closely the filter meets design specifications. To obtain a numerical listing of the frequency response, the FFT command must be used. This command requires that the filter's transfer function (the results of the DFI command) be stored as a record in a primary file. The file TQ100 is initialized as a record file and opened to accept record data by the OPN command.

Input: OPN

ILS responds with: (A system prompt.)

3. The DFI command is executed by entering the command with its arguments. The values of the arguments are determined using the design specifications of the problem. The order of the filter was determined to be two using non ILS techniques. For this problem, the P argument allows the filter to stored as a record. The first numeric argument is the order of the filter and the second is the passband ripple in milli-dB and must be 0 to indicate this is a Butterworth filter. The third argument is the integer sampling frequency in Hz, the fourth and fifth are the integer lower and upper band edges of the band pass filter. The sixth argument is the stop band edge or "dB down" and must be 0 for Butterworth filters. The last argument is the power of ten multiplier for the frequencies.

Input: DFI P2,0,500,25,36,0,2

ILS responds with: Figure 3.2, a listing of the transfer function of the filter.

4. To see what is stored as a record in the file TQ100, the LRE command with its default values is used.

Input: LRE

ILS responds with: Figure 3.3, a listing of the record data in file TQ100.

5. The FFT command is used to determine the frequency response of the filter. This command needs a secondary record file opened to store its results. The FIL command, with the S argument this time, selects file TQ 101 as a secondary file.

```
SAMPLING FREQUENCY       50000.000 HZ
BAND PASS BUTTERWORTH (MAXIMALLY FLAT) FILTER
BAND EDGES               2500.000     3600.000 HZ

       DENOMINATOR      NUMERATOR
  1    1.000000E+00    4.346089E-03
  2   -3.537423E+00     .000000E+00
  3    4.940204E+00   -8.692178E-03
  4   -3.207272E+00     .000000E+00
  5    8.224354E-01    4.346089E-03
                 POLES                          QUADRATIC FACTORS
           REAL          IMAGINARY      FIRST ORDER     SECOND ORDER
          .861813         .391486       -1.723626         .895984
          .906838         .308943       -1.813736         .917914
                 ZEROS                          QUADRATIC FACTORS
           REAL          IMAGINARY      FIRST ORDER     SECOND ORDER
       -1.000000          .000000        1.000000         .000000
        1.000000          .000000       -1.000000         .000000
       -1.000000          .000000        1.000000         .000000
        1.000000          .000000       -1.000000         .000000

TIME CONSTANT                23.350 SAMPLES
NOISE BANDWIDTH            1218.451 HZ
```

Figure 3.2 Output of DFI Command.

```
C:\ILS>LRE
*******************************************************************
* C:\ILS\TQ100.                                  100,       1  RECORDS *
*******************************************************************

-------------------------------------------------------------------
  RECORD       1, SAMPLING FREQUENCY 5.00E+04, TYPE 1121

REAL TIME SERIES OF FILTER COEFFICIENTS

  INDEX    NUMERATOR     DENOMINATOR

     1     4.3461E-03     1.0000E+00
     2      .0000E+00    -3.5374E+00
     3    -8.6922E-03      .9402E+00
     4      .0000E+00    -3.2073E+00
     5     4.3461E-03     8.2244E-01
ENTER C TO CONTINUE, E TO EXIT, N FOR NEXT RECORD
->C
RECORD       2 NOT FOUND
```

Figure 3.3 File TQ100. Record Data.

Input: FIL STQ101

ILS responds with: TQ101. DOES NOT EXIST

SECONDARY FILE

6. The OPN command is used as before to initialize TQ101 as a record file. The S argument must be used since TQ101 is a secondary file.

Input: OPN S

ILS responds with: (A system prompt.)

7. The FFT command is executed to determine the frequency response. Since the primary file contains the filter's transfer function, the command determines the frequency response of the filter by dividing the FFT of the numerator by the FFT of the denominator. By default, the order of the FFT used in each case, is the smallest that contains all data points and the record is automatically zero padded to bring it to the size of the FFT used. The arguments used with this command are P, which stores the FFT in polar format, and 9, which specifies that $2^9 = 512$ points are to be used.

Input: FFT P,,9

ILS responds with: TQ101. RECORD 1 STORED

8. The frequency response of the filter can now be displayed using the DRE command since the file TQ101 contains record data. The arguments used are M, which displayes only the magnitude portion of the data, and S, which tells DRE to display the secondary file.

Input: DRE MS

ILS responds with: Figure 3.4, the frequency response of the Butterworth filter.

9. To get a better picture, DRE is used with the A argument and the user is prompted for plotting and scaling options. The options allow the user to erase the screen before display, display with or without a grid, change the scale of the axes, enter labels for the display, overlay plot, offset the plot, and provide other plotting options.

Input: DRE MAS

ILS responds with: Scaling options (figure 3.5) and the scaled response (figure 3.6).

The last part of the analysis is to check the results and reevaluate if necessary. The results are satisfactory and iteration is not required. [Ref. 4]

Problem 2: Using the bilinear transform and the design specifications of problem (1), design a minimum order Chebyshev filter and plot the frequency response [Ref. 3: p. 714].

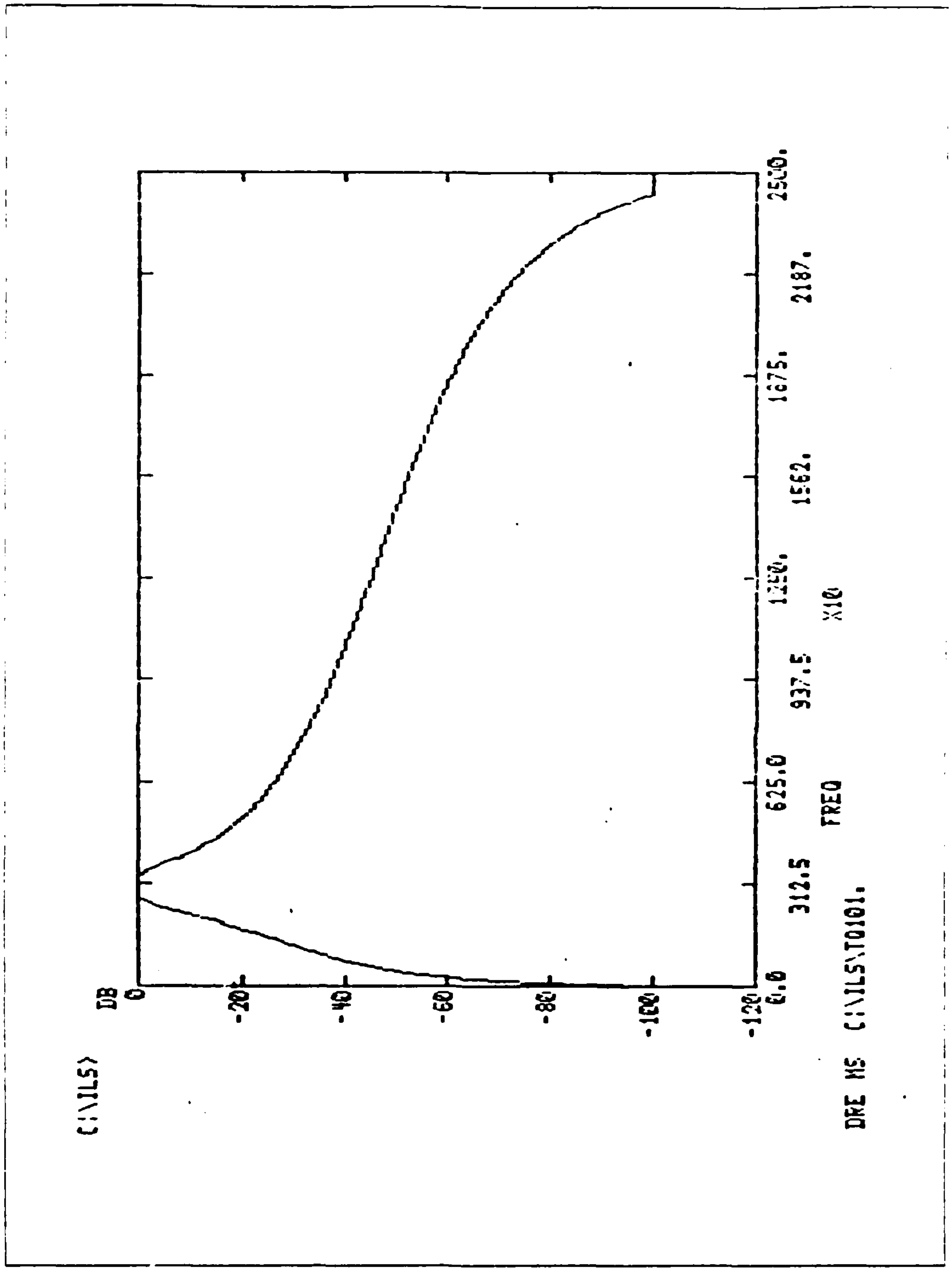

Figure 3.4 Frequency Response of Butterworth Filter.

```
->DRE MA3
PLEASE ENTER ERASE OPTION :
N : NOT ERASE BEFORE DISPLAY
DEFAULT : ERASE
->
PLEASE ENTER GRID OPTIONS :
N : NO GRID
C : CROSSHATCH GRID
DEFAULT : TICK MARK GRID
->C
PLEASE ENTER SCALING OPTIONS :
M : MANUAL INPUT MIN. MAX. SCALING
R : REPEAT PREVIOUS MIN. MAX.
DEFAULT : AUTOMATIC SCALING
->M
THE PREVIOUS SCALING :
    XMIN,       XMAX,      Y1MIN,     Y1MAX,     Y2MIN,      Y2MAX
 .000       .250E+05 -120.       .000       .000       .000
PLEASE ENTER NEW VALUES
=>2200.,4000.,-7.5,0.5,0.,0.
PLEASE ENTER LABEL OPTIONS :
N : NO LABELS
M : MANUAL INPUT LABELS
R : REPEAT PREVIOUS LABELS
DEFAULT : AUTOMATIC LABELS
->
PLEASE ENTER PROFILE OPTIONS :
O : OVERLAY PLOT
A[P,N,A,B] : AUTOMATIC OFFSET
R[P,N,A,B] : REPEAT PREVIOUS OFFSET
M[P,N,A,B] : MANUAL INPUT OFFSET
     [P,N,A,B] : VERTICAL BARS FOR POS. NEG, ALL OR BOTTOM
DEFAULT : AUTOMATIC OFFSET PROFILE PLOT
->
PLEASE ENTER TYPE OPTIONS : [D][XY][CC,CL,LC,LL]
->
```

Figure 3.5 ILS Scaling Option Prompts.

The method of analysis is similar to the previous problem except in this case ILS is used to determine the order of the filter. The commands used are the same, however, the COMMON file will be used to store intermediate results. A new command, FDI, is used to display the frequency response of the filter using the filter coefficients stored in the COMMON file. Once the design specifications are met, the filter coefficients of the COMMON file are transferred to a record file for further analysis. The ILS commands used in this procedure are

- Design Filter Command (DFI).
- Frequency Display Command (FDI). This command computes and displays a frequency spectrum.
- File Command (FIL).

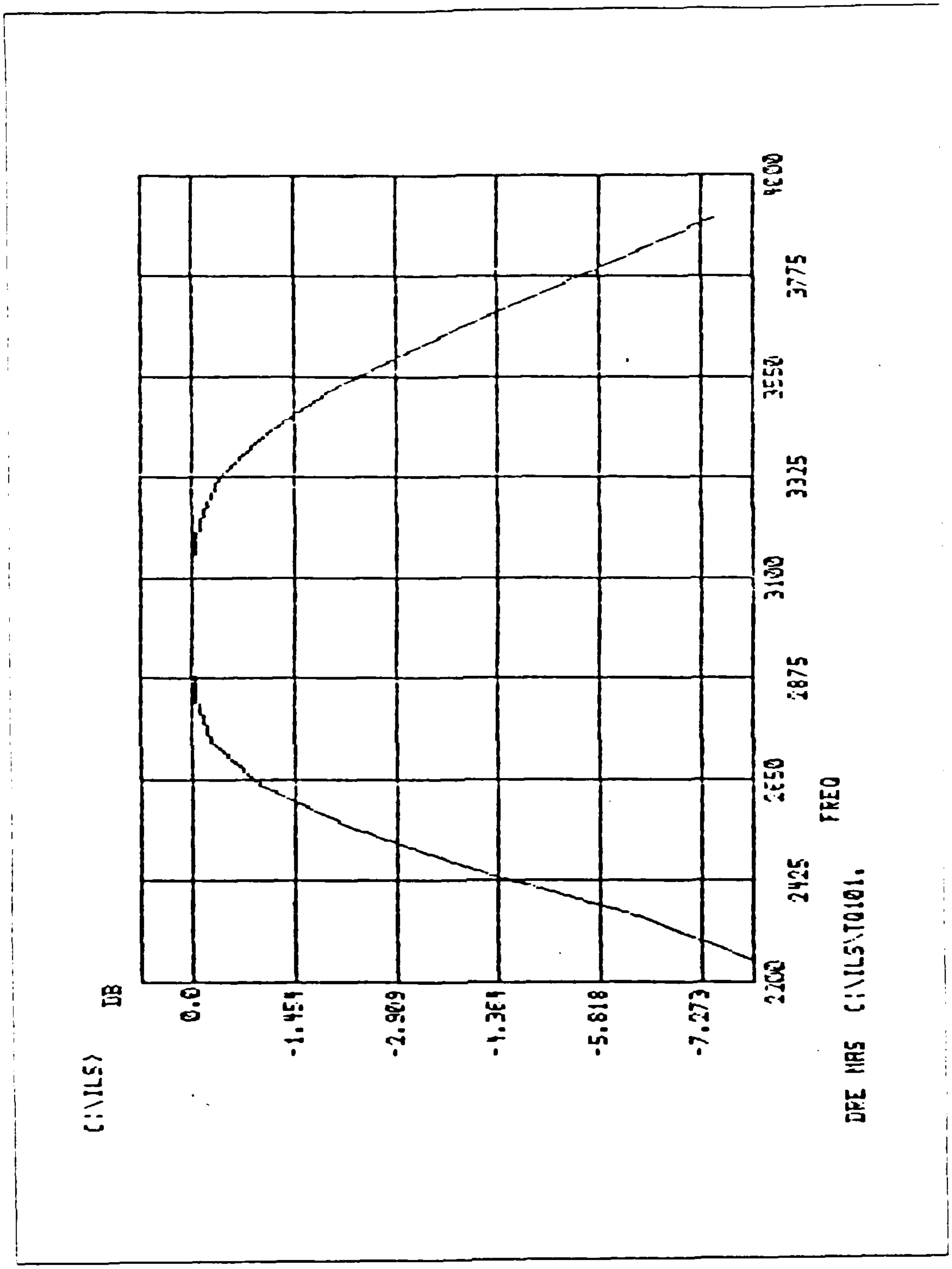

Figure 3.6 Scaled Frequency Response.

- Store Records Command (SRE). This command stores sampled data as records.
- Fast Fourier Transform Command (FFT).
- Display Record Command (DRE).

The design sequence is as follows:

1. Set the alphabetic prefix for the data files to be used in this analysis.

Input: FIL ANTQ

ILS responds with: Alphabetic characters set to: TQ

2. The DFI command is used interactively to design the Chebyshev filter. ILS prompts for inputs relating to the specifications of the filter to be designed and determines the transfer function of the filter which is stored in the COMMON file.

Input: DFI

ILS responds with: Figure 3.7, the ILS prompts, for filter design specifications, caused by interactive use of the DFI command. The -> and => symbols are prompts for user input which are determined using the design specifications. The filter specifications as determined by responding to the DFI prompts are displayed in figure 3.8.

3. The frequency response of the filter is examined using FDI. This command automatically computes and displays the frequency response of the filter stored in the COMMON file. The E argument erases the terminal screen before displaying the frequency response and the G argument places a grid over the display. The C argument tells the FDI command to use an FFT to compute the frequency spectrum of a rational form filter stored in the COMMON file.

Input: FDI EGC

ILS responds with: Figure 3.9, the frequency response of the filter in the COMMON file.

4. The frequency response of the filter in the passband region is not easily determined from the display because more resolution is needed. This is done with the DRE command. In order to use this command the filter data in the COMMON file must be moved to a record file so the frequency response can be recomputed and stored in a file type which the DRE command can use as in Problem (1). The SRE command has a special argument that allows the

```
C:\ILS>DFI

1 = LOW-PASS, 2 = BAND-PASS, 3 = HIGH-PASS, 4 = BAND-REJECT
PLEASE ENTER NUMBER
->2

ENTER SAMPLE FREQUENCY, PASS BAND EDGE(S) (FLT PT)
=>50000.,100   2800.,3210.

ENTER STOP BAND EDGE, ONLY ONE MAY BE USER SPECIFIED,
EVEN WHEN FILTER IS PASS BAND OR STOP BAND (FLT PT)
=>1000.


STOP BAND EDGE(S)  1000.00  8337.94

TRANSITION RATIO=19.28, THETA= 2.97

ENTER PASS BAND VARIATION (MILLIDB), AND STOP BAND ATTENUATION (DB) (FLT PT)
WITH CHEBYCHEV AND ELLIPTIC FILTERS THERE WILL BE RIPPLES IN THE
PASS BAND. WITH THE ELLIPTIC AND CHEBYCHEV II THERE WILL BE
RIPPLES IN THE STOP BAND. 0 DB DEFAULTS TO 3.0103 DB.

PLEASE ENTER THE TWO REQUESTED NUMBERS (FLT PT)
=>500.,20.
MINIMUM PROTOTYPE ORDER REQUIRED (NOTE THAT FOR THE DIGITAL PASS BAND
OR STOP BAND FILTERS, THIS WILL BE DOUBLED):

BUTTERWORTH  1.1, CHEBYCHEV  1.1, ELLIPTIC  1.1

ENTER ORDER OF PROTOTYPE FILTER YOU WISH TO USE
->2

ENTER NUMBER TO INDICATE WHICH PARAMETER TO CHANGE
1 = STOP BAND EDGE,  2 = DB ATTENUATION
->1
PASS BAND EDGE(S)                    2800.0   3210.0
BUTTERWORTH HALF POWER POINT(S)      2670.7   3364.1
STOP BAND EDGE(S) ELLIPTIC           2309.8   3879.2
STOP BAND EDGE(S) CHEBYCHEVS             .0       .0
STOP BAND EDGE(S) BUTTERWORTH        2093.1   4268.1

THE FOLLOWING LIST THE ENTRIES YOU COULD MAKE DIRECTLY ON THE
COMMAND LINE TO DESIGN THE FILTERS AND AVOID ANY PROMPTING

DFI      2.     0.  5000.   267.   336.     0 ->  BUTTERWORTH
DFI      2.   500.  5000.   280.   321.     0 ->  CHEBYCHEV
DFI      2.     0.  5000.     0.     0.   -20 ->  CHEBYCHEV TYPE II
DFI      2.   500.  5000.   280.   321.   -20 ->  ELLIPTIC

ENTER FILTER TYPE TO USE (THE PROGRAM WILL ENTER VALUES FOR YOU):

1 = BUTTERWORTH, 2 = CHEBYCHEV, 3 = INVERTED CHEBYCHEV, 4 = ELLIPTIC
->2
```

Figure 3.7 DFI System Prompts.

```
DFI      2   500  5000   280   321      0      1

SAMPLING FREQUENCY        50000.000 HZ
BND PASS CHEBYCHEV FILTER
PASS BAND RIPPLE                .500 DB
BAND EDGES              2800.000    3210.000 HZ

      DENOMINATOR      NUMERATOR
 1    1.000000E+00    9.161502E-04
 2   -3.649376E+00     .000000E+00
 3    5.258056E+00   -1.832300E-03
 4   -3.518317E+00     .000000E+00
 5    9.292045E-01    9.161502E-04
            POLES                          QUADRATIC FACTORS
        REAL          IMAGINARY       FIRST ORDER     SECOND ORDER
       .902044         .384655        -1.804083         .961644
       .922944         .338292        -1.845688         .966267
            ZEROS                          QUADRATIC FACTORS
        REAL          IMAGINARY       FIRST ORDER     SECOND ORDER
      -1.000000        .000000         1.000000         .000000
       1.000000        .000000        -1.000000         .000000
      -1.000000        .000000         1.000000         .000000
       1.000000        .000000        -1.000000         .000000

TIME CONSTANT                  58.283 SAMPLES
NOISE BANDWIDTH               610.162 HZ
```

Figure 3.8 Filter specifications.

transfer of this data from the COMMON file to a secondary record file. First, the FIL command must be used to initialize a secondary file.

Input: FIL SDEYTQ200

ILS responds with: TQ200. DOES NOT EXIST

SECONDARY FILE

5. The OPN command is used as before to initialize TQ200 as a record file. The S argument must be used since TQ200 is a secondary file.

Input: OPN S

ILS responds with: (a system prompt)

6. The SRE command transfers data from the common file to a secondary record file. The F argument allows the transfer of the infinite impulse response filter coefficients, stored in the COMMON file by executing DFI, to a secondary record file.

Input: SRE F

ILS responds with: TQ200. RECORD 1 STORED

7. LRE is used to list the transfered data stored in TQ200. The sampling frequency has been normalized but this should not affect the frequency response of the filter.

Input: LRE S

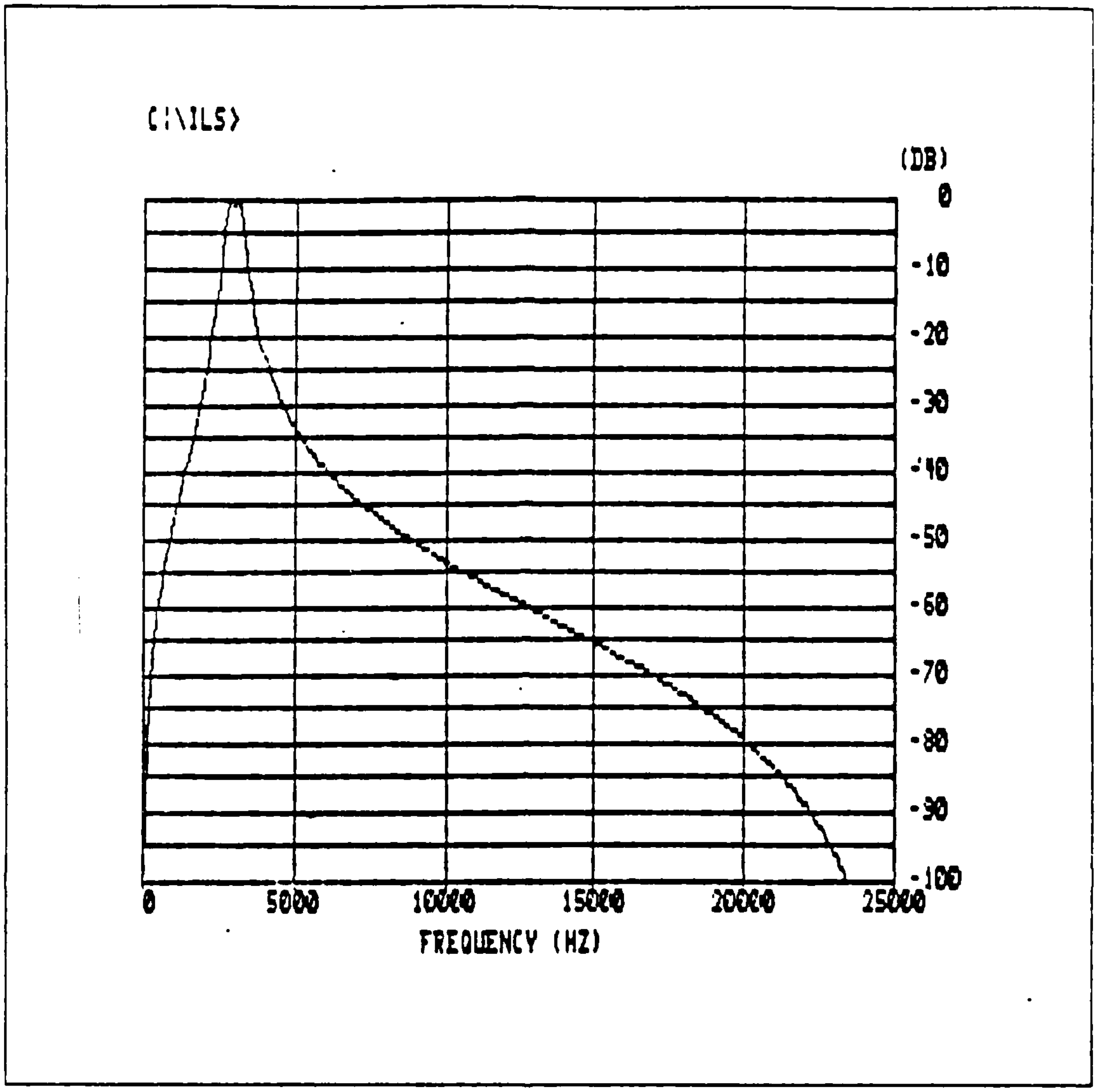

Figure 3.9 Frequency Response of Problem 2.

ILS responds with: Figure 3.10, the data from the COMMON file stored as a record in the secondary file TQ200.

8. The frequency response is computed using the FFT command and displayed using DRE. This FFT command requires making TQ200 a primary file and initializing a secondary record file to store the results of the FFT.

Input: FIL TQ200

ILS responds with: TQ200. RECORD 1 STORED
PRIMARY FILE

9. The FIL command is used to initialize a secondary file.

```
C:\ILS>DRE
*******************************************************************
* C:\ILS\TQ200.                                     200,        1 RECORDS *
*******************************************************************

-------------------------------------------------------------------
  RECORD        1, SAMPLING FREQUENCY 5.00E+04, TYPE 1121

REAL TIME SERIES OF FILTER COEFFICIENTS

  INDEX    NUMERATOR     DENOMINATOR

     1     9.1615E-04     1.0000E+00
     2      .0000E+00    -3.6500E+00
     3    -1.8323E-03     5.2581E+00
     4      .0000E+00    -3.5183E+00
     5     9.1615E-04     9.2920E-01
ENTER C TO CONTINUE, E TO EXIT, N FOR NEXT RECORD
->
RECORD       2 NOT FOUND
```

Figure 3.10 Record Data, TQ200.

Input: FIL SDEYTQ201

ILS responds with: TQ201. DOES NOT EXIST
SECONDARY FILE

10. The OPN command is used to initialize TQ201 as a secondary record file.

Input: OPN S

ILS responds with: (a system prompt)

11. The FFT command is executed to determine the frequency response.

Input: FFT P,,10

ILS responds with: TQ201. RECORD 1 STORED

12. The frequency response is displayed using DRE MS (figures 3.11) and DRE MAS (figure 3.12).

The filter is well within the design specifications. When examining the figures, note that the frequency axis has been normalized with respect to the sampling frequency. This is the result of transfering the filter from the COMMON file. Notice the straight line approximations that appear in figure 3.12 caused by the relatively small number of points computed in the passband. [Ref. 4]

Problem 3: Using the Fourier Series approach, design a non-recursive high pass filter of minimum order that meets the specifications of figure 3.13. The unshaded regions represent the desired response characteristic and the

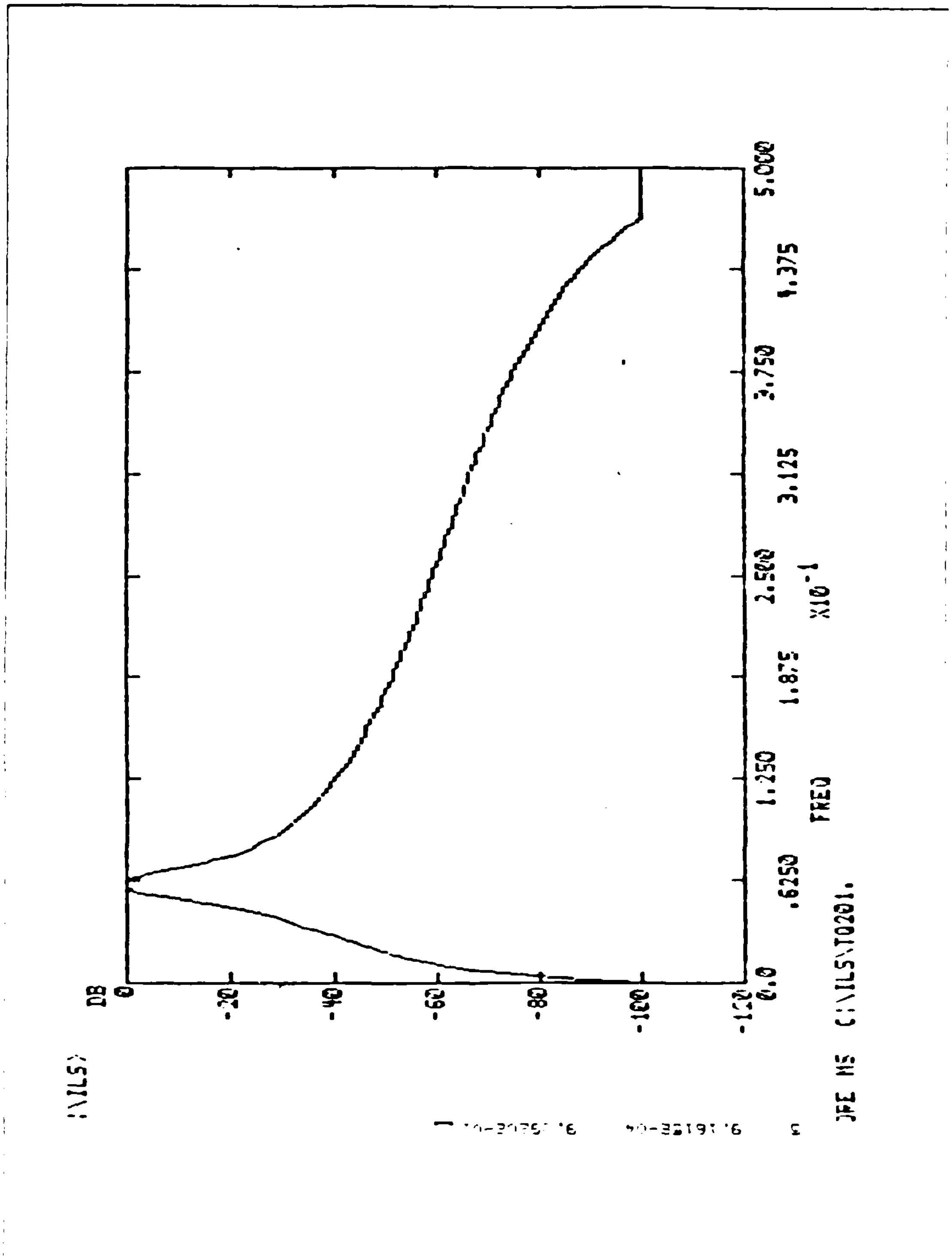

Figure 3.11 Frequency Response, Problem 2.

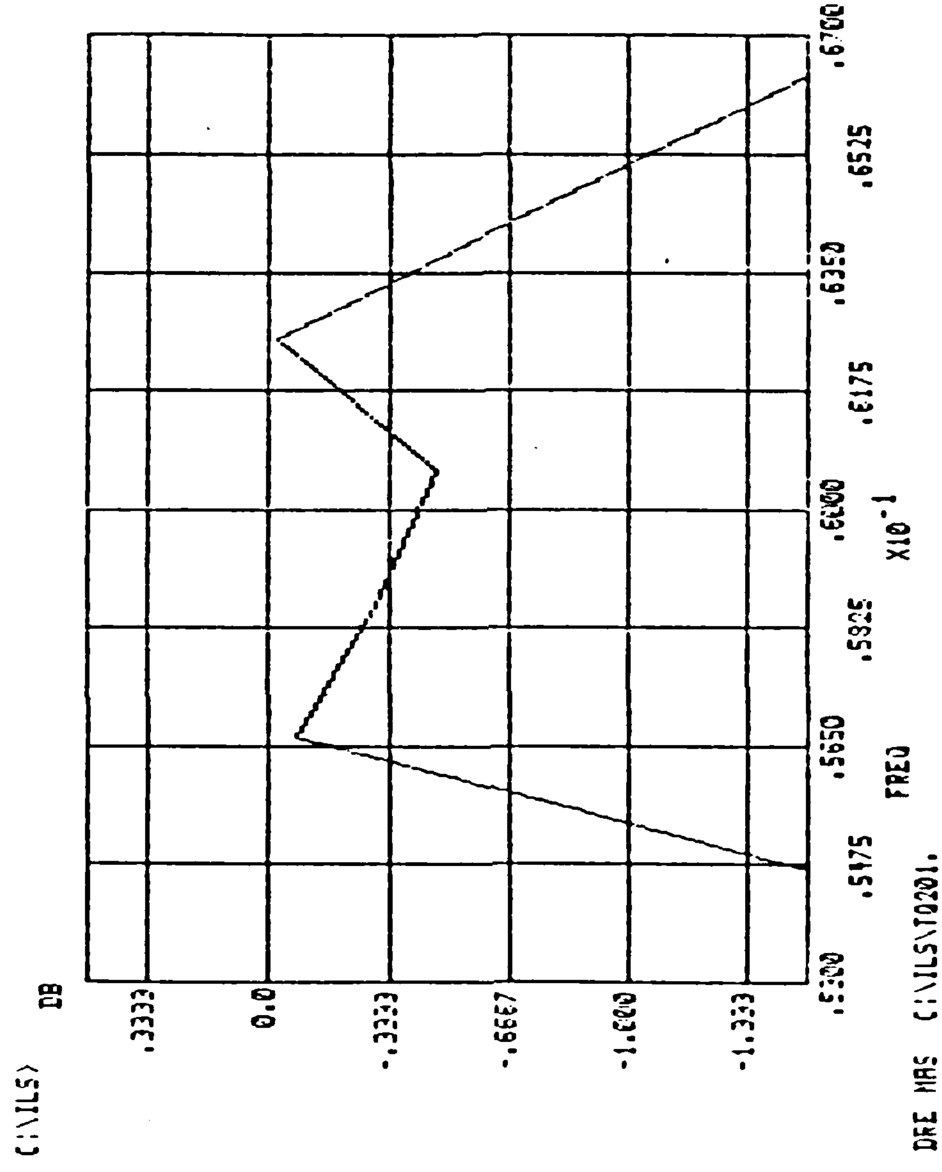

Figure 3.12 Frequency Response, Problem 2.

sampling frequency is 10K Hz. Plot the frequency response. [Ref. 3: p. 606].

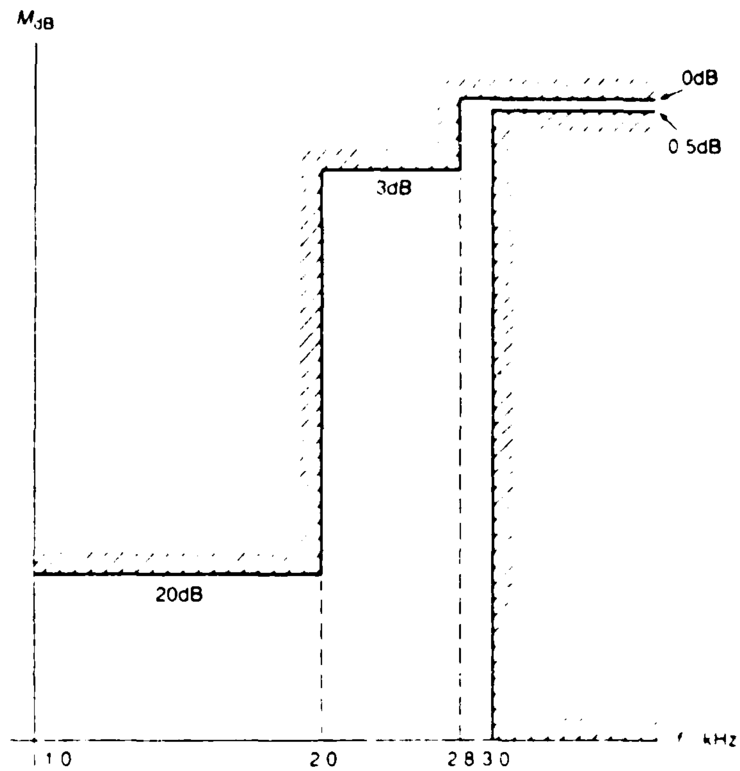

Figure 3.13 Specifications of Problem 3 [Ref. 3: p. 606]

This problem requires finding the unit sample response of the filter and truncating this with a windowing technique. Once the coefficients of the filter are determined, the frequency response can be plotted. The ILS commands used in this analysis are:

- File Command (FIL).
- Open Record File Command (OPN).
- Ideal Filter Design Command (IFL). This command generates the time domain impulse response of a specified ideal filter.

- Fast Fourier Transform Command (FFT).
- Display Records Command (DRE).

The IFL command is used to determine the coefficients of the filter. The FFT command determines the frequency response using the filter coefficients. The design sequence is as follows:

1. Set the alphabetic prefix for the data files to be used in this analysis, create a file to store the record generated by the IFL command and open the file to accept record data.

Input: FIL ANRF

ILS responds with: Alphabetic characters set to: RF

Input: FIL DEYRF300

ILS responds with: RF300. DOES NOT EXIST
PRIMARY FILE

Input: OPN

ILS responds with: (a system prompt)

2. The IFL command is used to determine the coefficients of the filter. This is done by using an FFT and the user specified spectral characteristics of the ideal filter. The first argument of the command is the order of the FFT for the design. The second, third and fourth are the integer sampling, lower cutoff and upper cutoff frequencies of the filter. The fifth is the power of ten multiplier for the frequencies, the sixth is the number of filter points to output and the seventh is the window type to use in the analysis. In the first trial of the design process, 31 coefficients are determined and a Hamming window is used by default.

Input: IFL 9,10000,2800,5000,0,31

ILS responds with: RF300. RECORD 1 STORED

3. A secondary file is created and opened to store the record data that will be generated by determining the frequency spectrum of the filter using the FFT command.

Input: FIL SDEYRF301

ILS responds with: RF301. DOES NOT EXIST
SECONDARY FILE

Input: OPN S

ILS responds with: (a system prompt)

4. The FFT command is used to determine the frequency response.

Input: FFT P,,10

ILS responds with: RF301. RECORD 1 STORED

5. The frequency response is displayed using the DRE MS (figure 3.14) and DRE MAS (figure 3.15) commands.

The filter does not meet the design specifications with 31 coefficients. After further iterations the specifications are meet using a filter of 57 coefficients (figures 3.16 and 3.17). This design uses more coefficients than necessary since the -3 dB cutoff frequency is approximately 2875 Hz and only 2800 Hz is required. Decreasing the cutoff frequency, the order of the filter can be further reduced. The final filter has an order of 41 coefficients (figures 3.18 and 3.19). [Ref. 4]

B. SPECTRAL ANALYSIS

In the following examples ILS will be used to perform spectral analysis. In the fist problem, ILS is used to compute the DFT of two finite sequences. In the second problem ILS is used to perform the convolution of two sequences. The last problem is also a DFT computation, however, the data for the sequence are input using an external data file.

Problem 4: Find the discrete Fourier transform for each of the following 40-point sequences [Ref 2: p. 486]:

a) $x_p(n) = 1, n = 0, 1, 39$
$x_p(n) = 0$, otherwise

b) $x_p(n) = 1, n = 0, 1, 2, 3, 37, 38, 39$
$x_p(n) = 0$, otherwise

In this problem ILS must be used to create the sequences and then find their DFTs. The commands to be used are:

- Context Command (CTX). This command lists or changes the context (number points per frame), sector number, or header length of a file.
- File Command (FIL).
- Initialize Command (INA). This command initializes or changes the primary sampled data file header.
- Modify Command (MDF). This command modifies the values of the sampled data or signal processing record data.
- Print Command (PRT). This command prints sampled data.
- Display Command (DSP). This command displays a time series waveform on the screen.

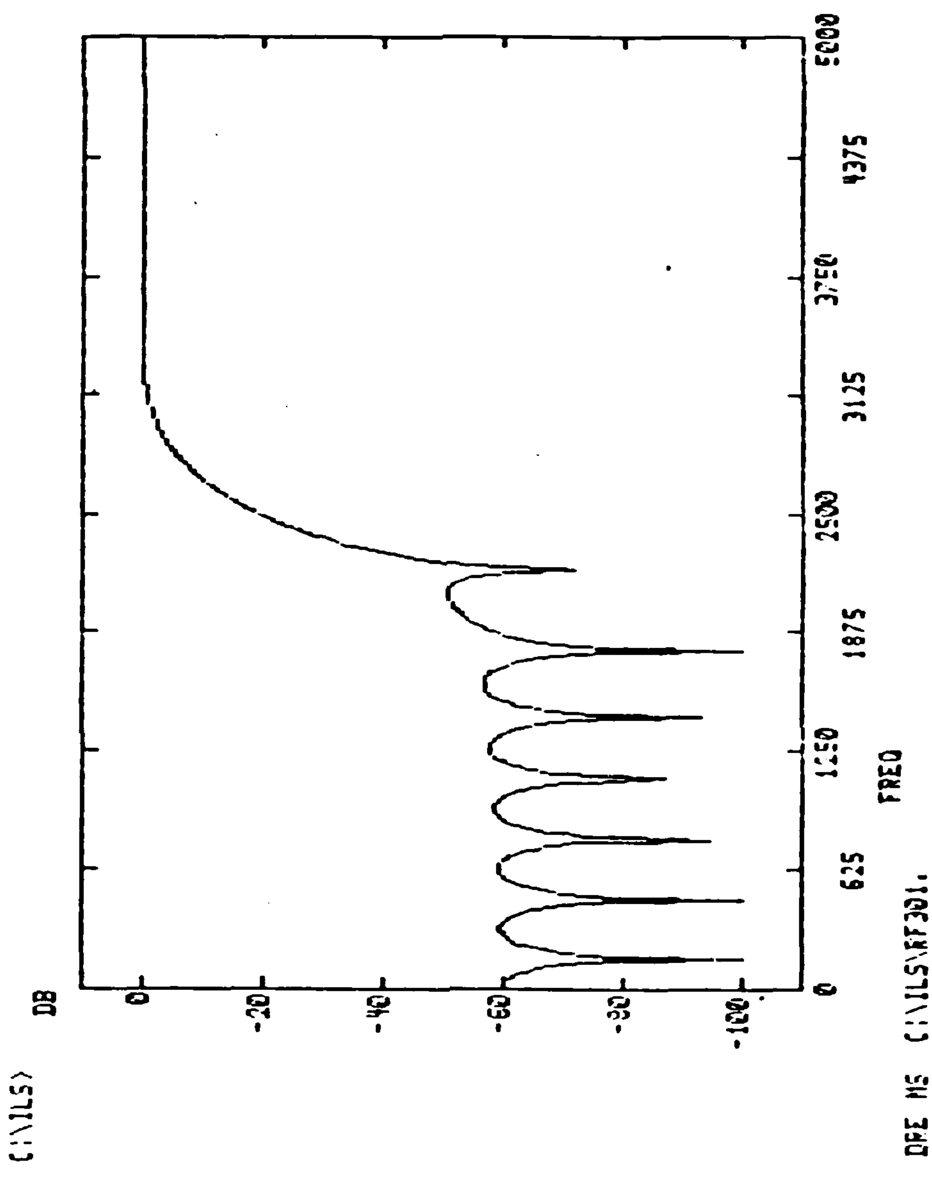

Figure 3.14 Frequency Response, 31 Coefficients.

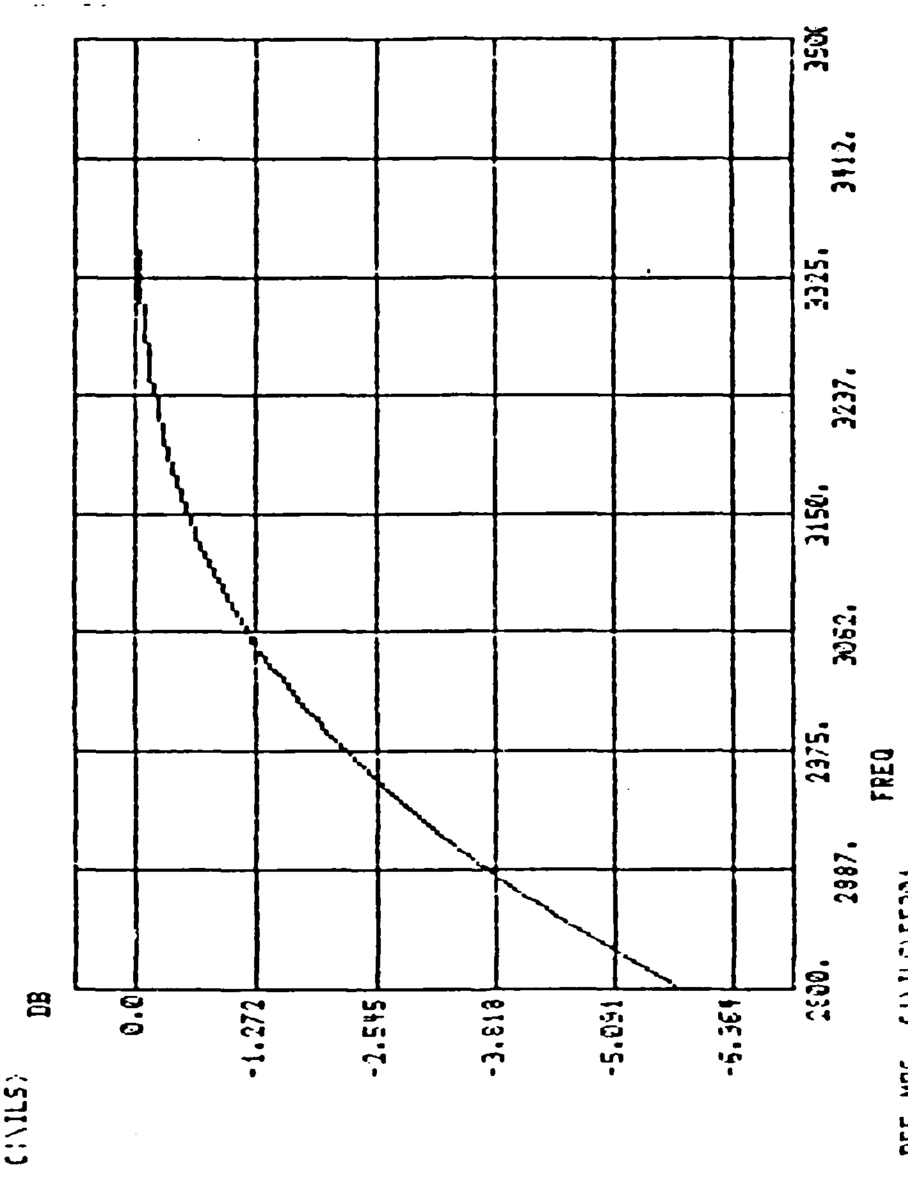

Figure 3.15 Frequency Response, 31 Coefficients.

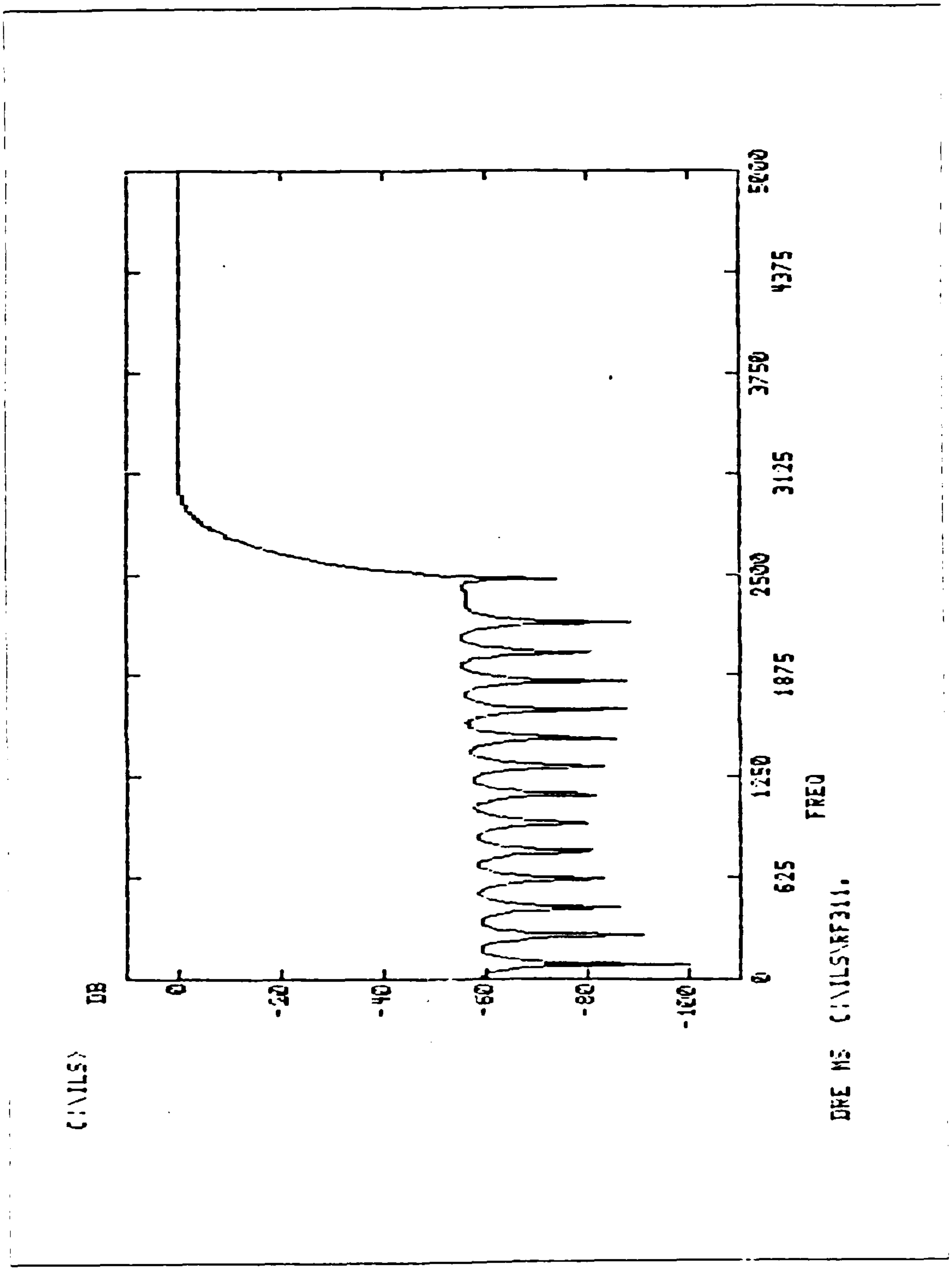

Figure 3.16 Frequency Response, 57 Coefficients.

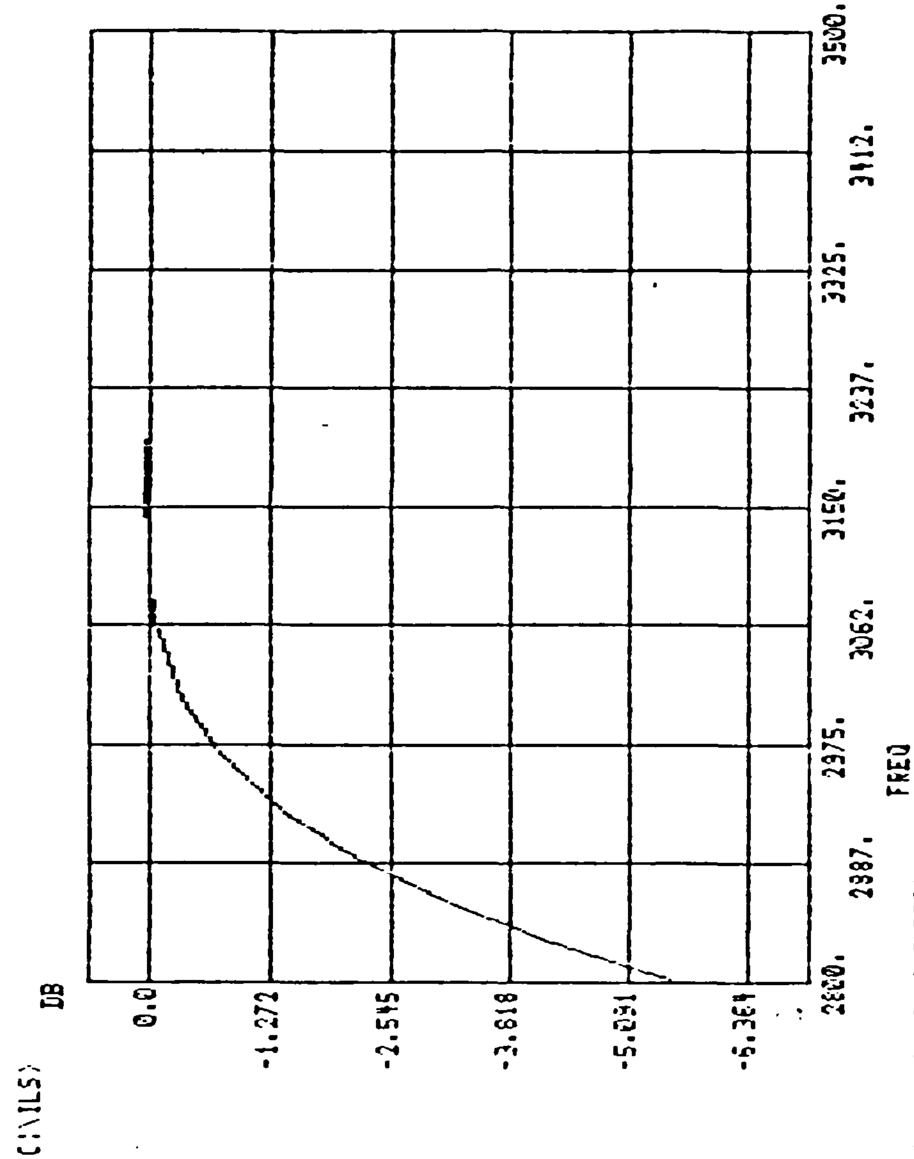

Figure 3.17 Frequency Response, 57 Coefficients.

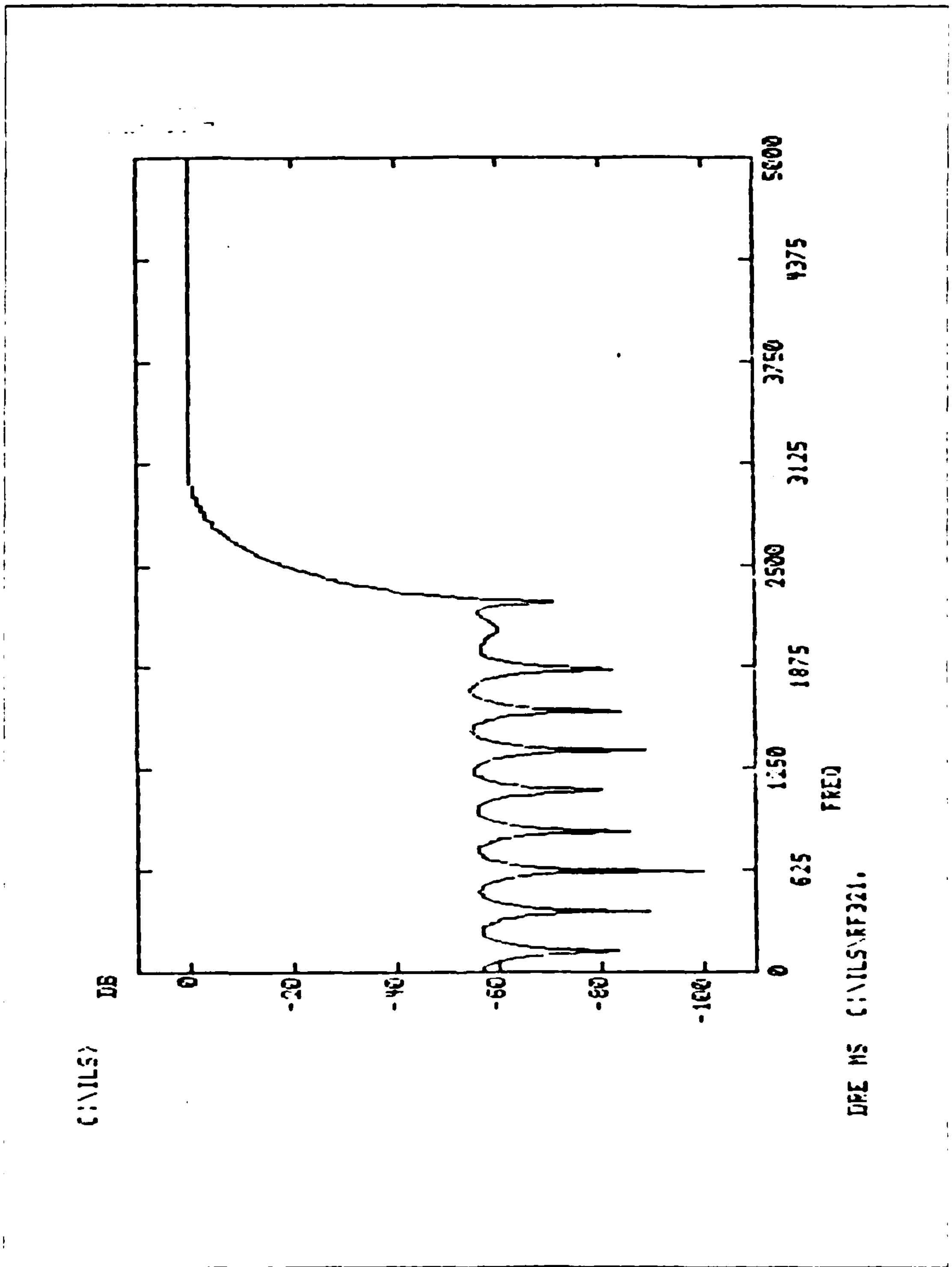

Figure 3.18 Frequency Response, 41 Coefficients.

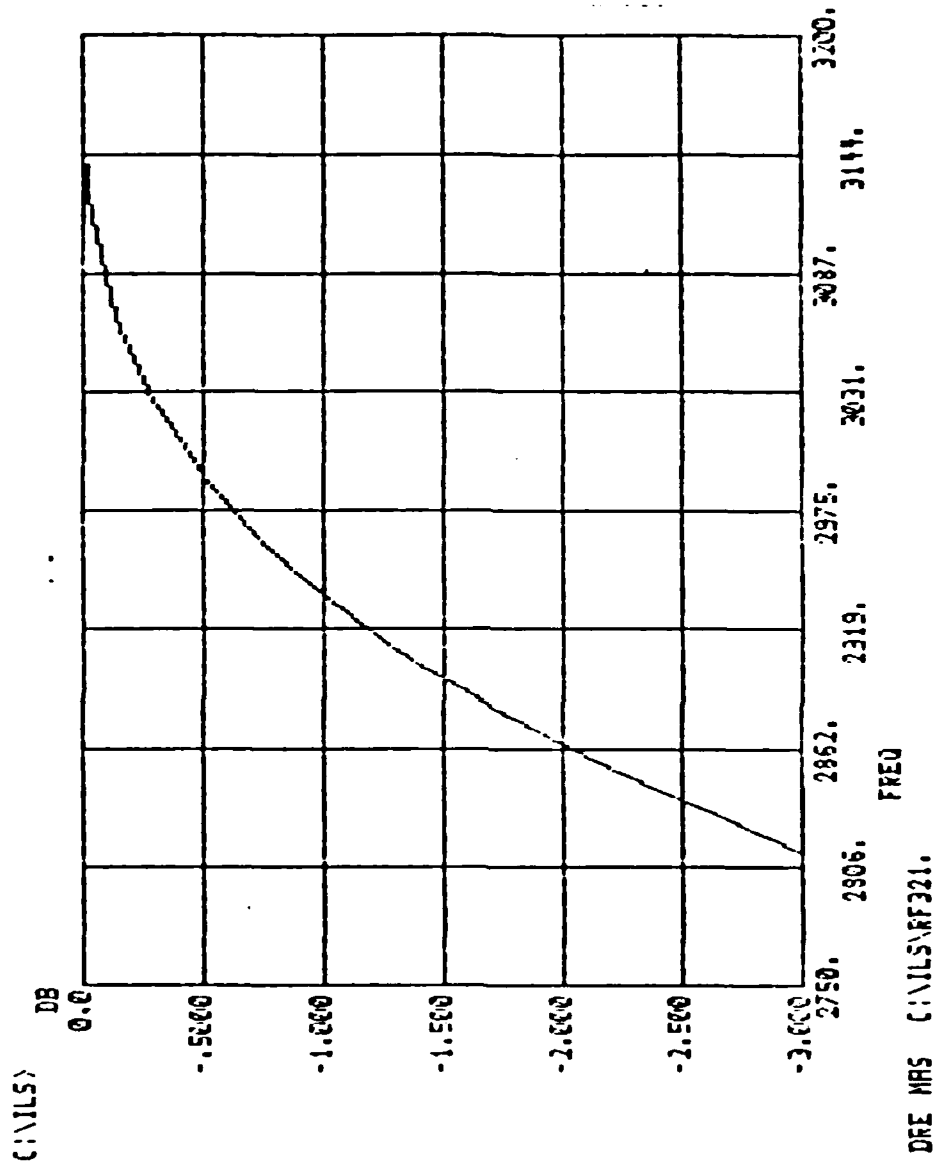

Figure 3.19 Frequency Response, 41 Coefficients.

• Frequency Display Command (FDI).

The sequence of the analysis is as follows:

1. The number of points per frame that a sampled data file contains is designated by the CTX command. In this problem, the sequences are 40-points. FDI determines the DFT of a sampled data file by frame and therefore, to obtain a DFT with no discontinuities the context, of the sampled data files must be set to 40-points per frame, the implicit period of the sequence.

Input: CTX 40

ILS responds with: CONTEXT = 40 POINTS FRAME

2. The prefix of the data files to be used in the analysis is initialized using the FIL command with the AN argument.

Input: FIL ANDF

ILS responds with: Alphabetic characters set to: DF

3. The FIL command is used to create a file of 40 zeros. The first argument, CRZ, creates a file filled with zeros. The numeric arguments of the command determine the size and number of files created. The first argument is the filename and filename numeric which identifies the primary file, the second declares the number of files to create and the third declares the number of frames per file to create. In this problem, two files are created each with one frame. Only one filename numeric is entered since ILS sequentially numbers the second file.

Input: FIL CRZDF100,,2,1

ILS responds with: DF100. NOT SAMPLED DATA
2 DK BLKS

DF101. NOT SAMPLED DATA
2 DK BLKS

DF100. NOT SAMPLED DATA
2 DK BLKS
PRIMARY FILE

4. In order for this file to be sampled data, a sampling frequency must be assigned to the file. ILS commands do not perform analysis with files unless a file type is declared. For example, assigning a sampling frequency makes the

file a sampled data file, opening the file makes the file a record file. Examining the problem, a sampling frequency of 40 Hz is assigned to the file. This gives the DFT a frequency spacing of 1 Hz because there are 40 samples and this yields a record length of 1 second. The sampling frequency of the file is assigned using the INA command with the argument SF. This argument allows INA to set the sampling frequency of the file making it a sampled data file. The first numeric argument used with the command is the integer multiple of the sampling frequency and the second is the power of ten multiplier for the sampling frequency.

Input: INA SF4,1

ILS responds with: SF = 40 SAMPLING FREQUENCY

5. The same must be done for file DF101. First the FIL command is used to make DF101 a primary file and then the INA command can be used to assign the sampling frequency.

Input: FIL DF101

ILS responds with: DF101. NOT SAMPLED DATA
2 DK BLKS
PRIMARY FILE

Input: INA SF4,1

ILS responds with: SF = 40 SAMPLING FREQUENCY

6. The sampled data files, which consist of zeros, are modified one at a time using the MDF command. First the FIL command identifies the file to be modified. The MDF command is used to change the elements of the file to represent the sequence. The first argument is I, which allows the user to modify unscaled integer values of sampled data. The first numeric argument identifies the starting frame of the file to modify, the second identifies the starting point in the frame to modify, the third is the data value which replaces the old value, the fourth is the number of consecutives frames to modify and the fifth is the number of consecutive points to modify. The first modification is in the first frame starting with the first point in the file. The zero is changed to a one and two consecutive points are altered.

Input: FIL DF100

ILS responds with: FIL DF100. SAMPLED DATA

2 DK BLKS. 1. FRAMES. 40 PT FR

SAMPLE RATE = 40 HZ

PRIMARY FILE

Input: MDF 11,1,1,1,2

ILS responds with: OLD = 0. NEW = 1

Input: MDF 11,40,1,1,1

ILS responds with: OLD = 0. NEW = 1

7. Once the modification is completed, the PRT command is used to display the data and check for correctness. The first numeric argument of PRT identifies which frame in the file to start displaying and the second declares the number of frames in the file to display.

Input: PRT 1,1

ILS responds with: Figure 3.20, the values of the sampled data file DF100.

```
C:\ILS>PRT 1,1
SECTOR  1, FRAME        1
      1      1      0      0      0      0      0      0      0      0
      0      0      0      0      0      0      0      0      0      0
      0      0      0      0      0      0      0      0      0      0
      0      0      0      0      0      0      0      0      0      1
C:\ILS>
```

Figure 3.20 Sampled Data File, DF100.

8. A similar process is repeated to change and display the other file, DF101.

Input: FIL DF101

ILS responds with: FIL DF101. SAMPLED DATA

2 DK BLKS. 1. FRAMES. 40 PT FR

SAMPLE RATE = 40 HZ

PRIMARY FILE

Input: MDF 11,1,1,1,4

ILS responds with: OLD = 0. NEW = 1

Input: MDF 11,38,1,1,3

ILS responds with: OLD = 0. NEW = 1

Input: PRT 1,1

ILS responds with: Figure 3.21, the values of the sampled data file DF101.

```
C:\ILS>PRT 1,1
SECTOR  1, FRAME     1
     1     1     1     1     0     0     0     0     0     0
     0     0     0     0     0     0     0     0     0     0
     0     0     0     0     0     0     0     0     0     0
     0     0     0     0     0     0     0     1     1     1
C:\ILS>
```

Figure 3.21 Sampled Data File, DF101.

9. The sequences, having been created and stored in sampled data files, can be displayed on a time axis using the DSP command. In this case, because the sequences are consecutive files, they can be displayed on the terminal screen simultaneously. First the FIL command is used to declare the first of the consecutive files as primary. The DSP command is then used to display the files. The argument E erases the terminal screen before plotting the data. The first numeric argument identifies the starting frame of the file to display, the second declares the number of frames to display, the third is the scaling factor for the data, the fourth makes the plot discrete, and the last identifies the number of consecutive files to plot. In this case, the plot consists of the first frame of files DF100 and DF101 and the data is scaled by 200.

Input: FIL DF100

ILS responds with: FIL DF100.   SAMPLED DATA
2 DK BLKS, 1. FRAMES, 40 PT FR
SAMPLE RATE = 40 HZ
PRIMARY FILE

Input: DSP E1,1,20000,2,,2

ILS responds with: Figure 3.22, a plot of the consecutive sampled data files.

10. The DFT of each sequence is computed using FDI. This command automatically plots the DFT as magnitude (dB) versus frequency. The command prompts the user for the frequency limits of the plot. If no limits are entered, ILS will plot the entire spectrum to the folding frequency. The command is used with the following arguments. The alphabetic argument, E

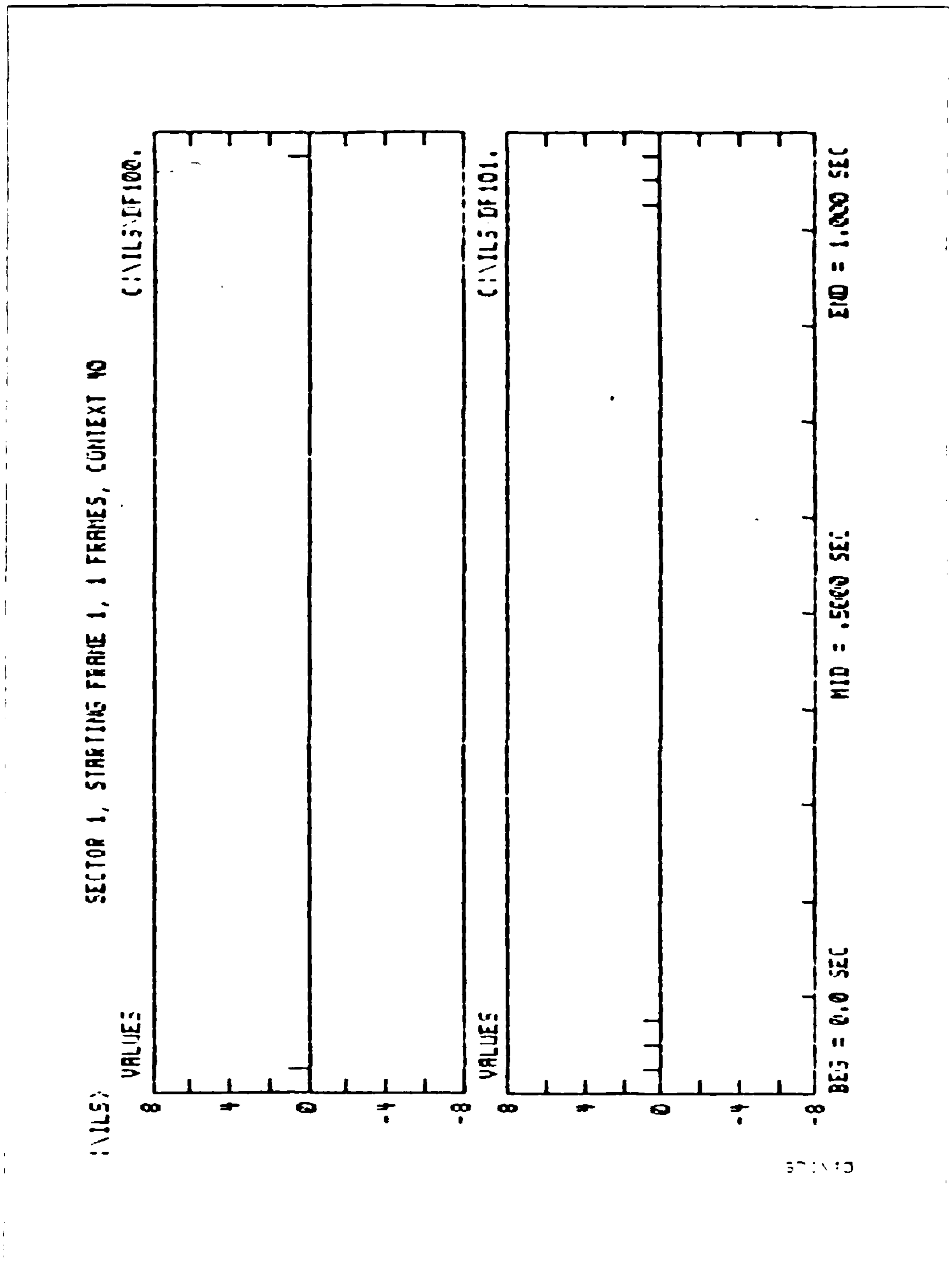

Figure 3.22 Plot of the Sequences of (a) and (b).

erases the terminal screen before plotting. GB places a grid border over the frequency display, and X directs ILS to determine the DFT of the data in a primary sampled data file. (The FIL command must be used to select a file for analysis by making it a primary file.) The first numeric argument of FDI identifies which frame in the file to start computing the DFT with, the second identifies the number of frames to be analyzed and the last requests the plot to use dots to represent the data points.

Input: FIL DF100

ILS responds with: FIL DF100. SAMPLED DATA
2 DK BLKS, 1. FRAMES, 40 PT FR
SAMPLE RATE = 40 HZ
PRIMARY FILE

Input: FDI EGBX1,1,,1

ILS responds with: Figure 3.23, the DFT of sequence (a).

11. The same procedure is performed for DF101.

Figures 3.23 and 3.24 are the DFTs of (a) and (b) respectively. Notice that ILS represents the DFT of a sequence using a continuous plot of 512 points. The coefficients of the DFT must be interpreted using the appropriate frequency resolution, Δf, of the sequences. [Ref. 4]

Problem 5: Two three-point sequences $x_1(n)$ and $x_2(n)$ are shown in figure 3.25. Sketch and label the linear convolution of the two sequences. [Ref. 3: p. 482].

As in the previous problem, ILS must generate the sequences. The data will be sampled data, however; ILS can only perform convolution with record data. ILS can do this analysis by changing the sampled data to record data using the SRE command. The commands needed for this analysis are as follows:

- Context Command (CTX).
- File Command (FIL).
- Initialize Command (INA).
- Modify Command (MDF).
- Print Command (PRT).
- Open Record File Command (OPN).
- Store Records Command (SRE).
- List Records Command (LRE).

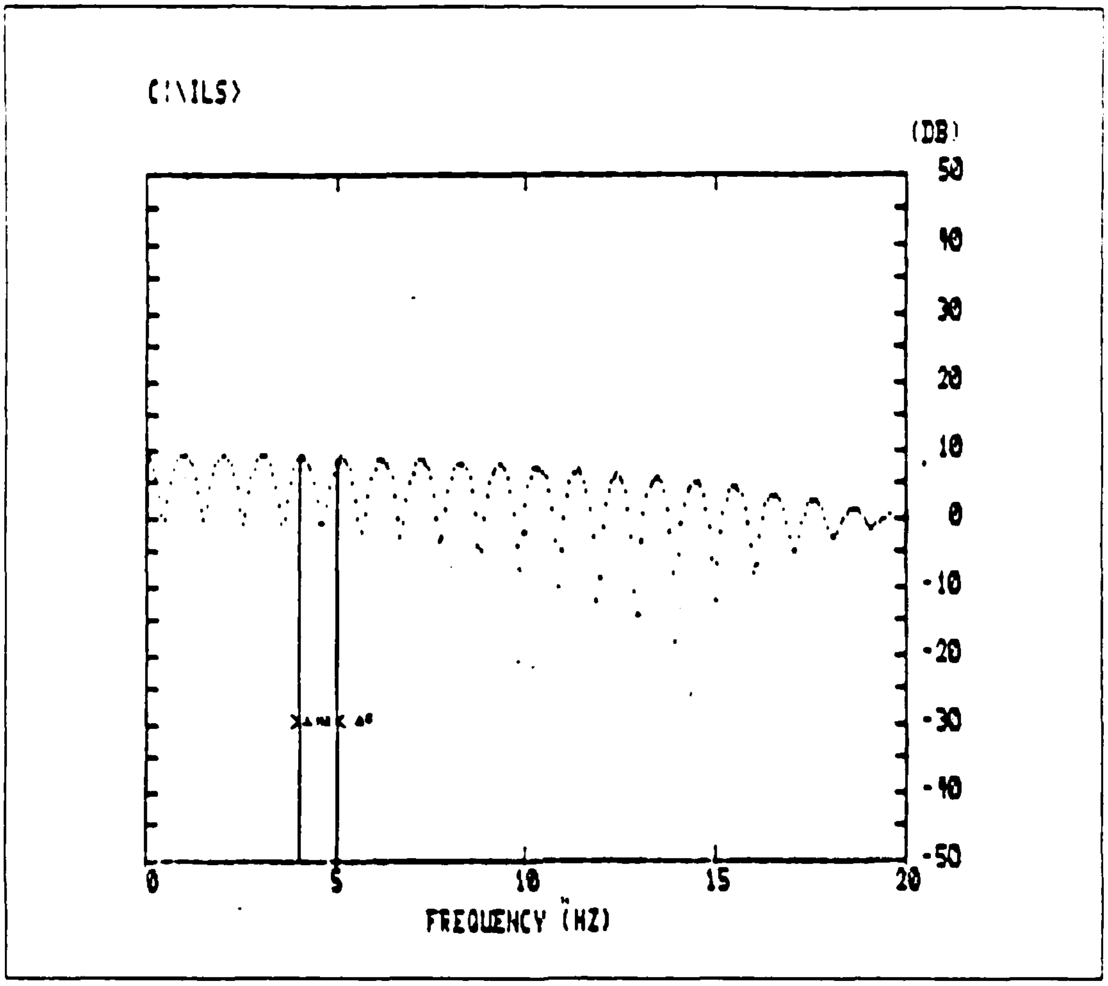

Figure 3.23 DFT of Sequence (a), $\Delta f = 1$ Hz.

- Convolution Command (CNV). This command convolves time series record data.
- Display Record Command (DRE).

The analysis is as follows:

1. Following a procedure similar to the previous problem, the sequences are generated. In this problem the files will be identified with the prefix WD. As before, the files initially contain zeros and are assigned an arbitrary sampling frequency to satisfy ILS requirements. The files are then modified with the MDF command to represent the sequences of figure 3.25. PRT is used to display the sequences and check for correctness. Since the CNV command

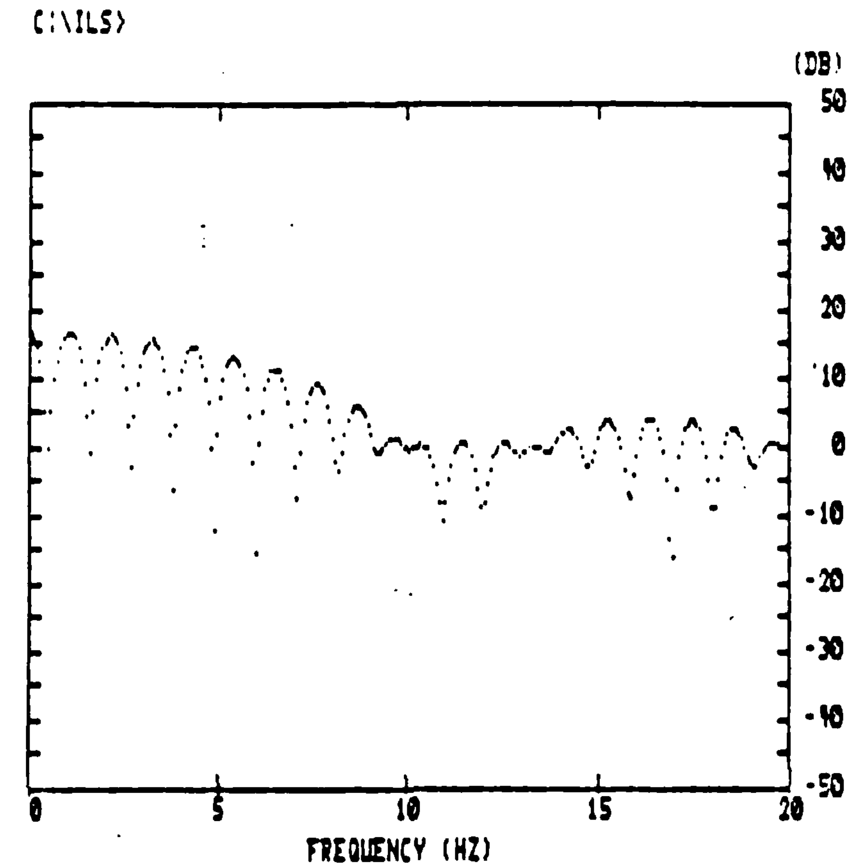

Figure 3.24 DFT of Sequence (b) , Δf = 1 Hz.

requires the files to be convolved be a power of 2, the context will be set to 8 points per frame. This context allows ILS to store the results of the convolution in another file of the same context. If the context is smaller, ILS would be unable to store the results of the convolution.

Input: CTX 8

ILS responds with: CONTEXT = 8 POINTS/FRAME

Input: FIL ANWD

ILS responds with: Alphabetic characters set to: WD

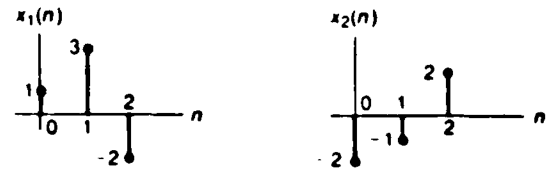

Figure 3.25 Sequences of Problem 2 [Ref 3: p. 482].

Input: FIL CRZWD100,,2,1

ILS responds with: WD100. NOT SAMPLED DATA
2 DK BLKS

WD101. NOT SAMPLED DATA
2 DK BLKS

WD100. NOT SAMPLED DATA
2 DK BLKS
PRIMARY FILE

2. With WD100 as the primary, a sampling frequency is assigned.

Input: INA SF1,0

ILS responds with: SF = 10 SAMPLING FREQUENCY

3. Now the data is modified one point at a time.

Input: MDF I1,1,1

ILS responds with: OLD = 0, NEW = 1

Input: MDF I1,2,3

ILS responds with: OLD = 0, NEW = 3

Input: MDF I1,3,-1

ILS responds with: OLD = 0, NEW = -2

4. The data is checked for correctness using PRT.

Input: PRT 1,1

ILS responds with: Figure 3.26, the sampled data point of file WD100.

5. The same procedure is performed for file WD101.

Input: FIL WD101

```
C:\ILS>PRT 1,1
SECTOR 1, FRAME 1
    1      3     -2      0      0      0      0      0
C:\ILS>
```

Figure 3.26 Sampled Data File WD100.

ILS responds with: WD101. NOT SAMPLED DATA

2 DK BLKS

PRIMARY FILE

Input: INA SF1,0

ILS responds with: SF = 10 SAMPLING FREQUENCY

Input: MDF I1,1,-2

ILS responds with: OLD = 0, NEW = -2

Input: MDF I1,2,-1

ILS responds with: OLD = 0, NEW = -1

Input: MDF I1,3,2

ILS responds with: OLD = 0, NEW = 2

6. The data is checked for correctness using PRT.

Input: PRT 1,1

ILS responds with: Figure 3.27, the sampled data points of file WD101.

```
C:\ILS>PRT 1,1
SECTOR 1, FRAME 1
   -2     -1      2      0      0      0      0      0
C:\ILS>
```

Figure 3.27 Sampled Data File WD101.

7. Now the sampled data of files WD100 and WD101 is made record data by duplicating the data as a records in the files WD200 and WD201 using SRE. This command requires the receiving files to be secondary files. This is

accomplished using the FIL command with the alphabetic argument S. The OPN command must also be used with the S argument to open these secondary files. SRE is used after the secondary record files are opened to receive the data as record data. The first numeric argument identifies the starting frame of the sampled data file and the second is the number of frames to duplicate.

Input: FIL WD100

ILS responds with: FIL WD100. SAMPLED DATA
2 DK BLKS, 1. FRAMES, 8 PT FR
SAMPLE RATE = 1.0 HZ
PRIMARY FILE

Input: FIL SWD200

ILS responds with: WD200. DOES NOT EXIST
SECONDARY FILE

Input: OPN S

ILS responds with: (a system prompt)

Input: SRE 1,1

ILS responds with: WD200. RECORD 1 STORED

Input: FIL WD101

ILS responds with: FIL WD101. SAMPLED DATA
2 DK BLKS, 1. FRAMES, 8 PT FR
SAMPLE RATE = 1.0 HZ
PRIMARY FILE

Input: FIL SWD201

ILS responds with: WD201. DOES NOT EXIST
SECONDARY FILE

Input: OPN S

ILS responds with: (a system prompt)

Input: SRE 1,1

ILS responds with: WD200. RECORD 1 STORED

8. LRE is used to check that the data has been duplicated successfully. Since the record files are secondary, the alphabetic argument S must be used. The first numeric argument in the command is the starting record of the files to list, the second identifies the number of records in each file to display and the third identifies the number of consecutive files to display. WD200 must also be the first secondary file identified by the command.

Input: FIL SWD200

ILS responds with: WD200. RECORD DATA
13 DK BLKS, 1 RECORDS
SECONDARY FILE

Input: LRE S1,1,2

ILS responds with: Figure 3.28, a listing of the record data in files WD200 and WD201.

```
C:\ILS>LER RE S1,1,2
*********************************************************************
* C:\ILS\WD200.                                   200,      1 RECORDS *
*********************************************************************

---------------------------------------------------------------------
  RECORD      1, SAMPLING FREQUENCY 1.00E+00, TYPE 1111

REAL TIME SERIES OF SAMPLED DATA

  INDEX        TIME          REAL VALUE

     1       .0000E+00     1.0000E+00
     2      1.0000E+00     3.0000E+00
     3      2.0000E+00    -2.0000E+00
     4      3.0000E+00      .0000E+00
     5      4.0000E+00      .0000E+00
     6      5.0000E+00      .0000E+00
     7      6.0000E+00      .0000E+00
     8      7.0000E+00      .0000E+00
ENTER C TO CONTINUE, E TO EXIT, N FOR NEXT RECORD
->C
*********************************************************************
* C:\ILS\WD201.                                   201,      1 RECORDS *
*********************************************************************

---------------------------------------------------------------------
  RECORD      1, SAMPLING FREQUENCY 1.00E+00, TYPE 1111

REAL TIME SERIES OF SAMPLED DATA

  INDEX        TIME          REAL VALUE

     1       .0000E+00    -2.0000E+00
     2      1.0000E+00    -1.0000E+00
     3      2.0000E+00     2.0000E+00
     4      3.0000E+00      .0000E+00
     5      4.0000E+00      .0000E+00
     6      5.0000E+00      .0000E+00
     7      6.0000E+00      .0000E+00
     8      7.0000E+00      .0000E+00
C:\ILS>
```

Figure 3.28 Record Data WD200 and WD201.

9. With the record files identified, CNV is used to perform convolution of the two sequences. The CNV command requires one file to be a Primary (A) file and the other to be a Primary (B) file. This is accomplished using the FIL command. Default declares a file Primary (A) and the alphabetic argument B declares a file Primary (B). CNV also requires a secondary file be opened to store the results of the convolution. This is accomplished as before using the FIL and OPN commands. WD300 is used to store the results. None of the CNV command's arguments are needed to perform the analysis.

Input: FIL WD200

ILS responds with: WD200. RECORD DATA
13 DK BLKS, 1 RECORDS
PRIMARY FILE

Input: FIL BCV201

ILS responds with: WD201. RECORD DATA
13 DK BLKS, 1 RECORDS
PRIMARY-B FILE

Input: FIL SWD300

ILS responds with: WD300. DOES NOT EXIST
SECONDARY FILE

Input: OPN S

ILS responds with: (a system prompt)

Input: CNV

ILS responds with: WD300. RECORD 1 STORED

10. The results can be displayed numerically using LRE or graphically using DRE. The file is secondary so the S argument must be used with either command. The M and A arguments used with DRE display the magnitude of the record while prompting for scaling options.

Input: LRE S

ILS responds with: Figure 3.29, a listing of the results of the convolution of WD200 and WD201.

The result is displayed in figure 3.30 using DRE MAS. [Ref. 4]

Problem 6: Evaluate the DFT of the sequence found by sampling the analog signal:

$$f(t) = \frac{2f_0 \sin(2\pi f_0 t)}{2\pi f_0 t}$$

```
C:\ILS>LRE A S
******************************************************************
* C:\ILS\WD300.                               300,      1 RECORDS *
******************************************************************

------------------------------------------------------------------
  RECORD      1, SAMPLING FREQUENCY 1.00E+00, TYPE 1111

REAL TIME SERIES OF SAMPLED DATA

  INDEX        TIME          REAL VALUE

      1      .0000E+00     -2.0000E+00
      2     1.0000E+00     -7.0000E+00
      3     2.0000E+00      3.0000E+00
      4     3.0000E+00      8.0000E+00
      5     4.0000E+00     -4.0000E+00
      6     5.0000E+00      1.1921E-07
      7     6.0000E+00     -1.1921E-07
      8     7.0000E+00     -5.3605E-08
ENTER C TO CONTINUE, E TO EXIT, N FOR NEXT RECORD
->C
RECORD      2 NOT FOUND
C:\ILS>
```

Figure 3.29 Record Data, $x_1(n) * x_2(n)$.

and retaining only those values in the interval $-1.0 \leq t < 1.0$ s. The sampling frequency is 16 Hz and 32 samples are taken. Find and plot the magnitudes of the DFT coefficients for f_0 equal to [Ref 2: p. 486.]:

a) 0.5 Hz

b) 1.0 Hz

In this problem, the 32 samples of the sinc function must be determined and then used to find the DFT coefficients for varying values of f_0. Since ILS cannot generate the sampled values of the sinc function, the data is determined using a Fortran program and stored in an ASCII data file which is then transfered to ILS. ILS can then be used to perform the analysis and plot the results. The following ILS commands are used in the analysis:

- File Command (FIL).
- Open Command (OPN).
- Write Command (WRT). Write sampled data or record data from a user ASCII data file, or write sampled data to/from a coded ASCII file.
- List Records Command (LRE).
- Fast Fourier Transform Command (FFT).
- Display Records Command (DRE).

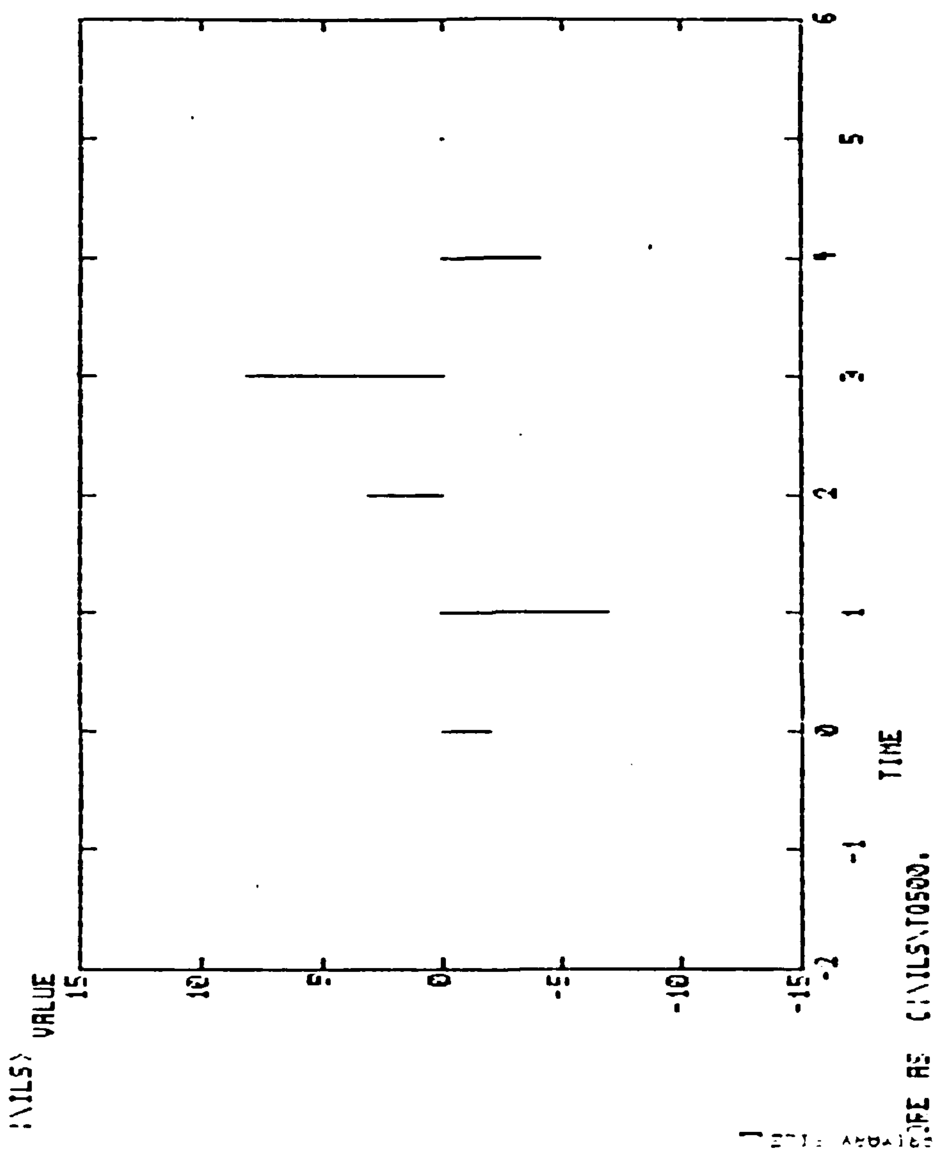

Figure 3.30 Record Data, $x_1(n) * x_2(n)$.

- Unary Operations Command (UOP). This command is used to perform unary operations on signal processing record files.

The first step, in this problem, is to write a program which generates the data required for the problem and stores the results in an ASCII data file. The data file must then be transfered to the ILS directory and a record file opened to store the data. The WRT command is used to write the data to ILS. This command requires the data to be in a specific format in a file named WRTIN.DAT. In this case, since the data is real-valued time series data, the format is F20.5 with one value per line. The program and the ASCII data file are displayed in figures 3.31 and 3.32 respectively. Before the WRT command is used, a file must be opened to accept the data. After the data is transfered, the FFT command is used to compute the DFT of the sequence. The FFT command is used instead of the FDI command because the number of elements in the sequence is a power of two. The sequence of ILS commands are as follows:

```
C       THIS PROGRAM GENERATES DATA FOR ILS.
        REAL X(32), Z, PI, T
        OPEN (UNIT=9, FILE='WRTIN.DAT', STATUS='OLD')
C        DIMENSION X(600)
        N=32
        PI=ACOS(-1.0)
        DO 20 I=1,N
              Z=I-1.0
              T=(Z*0.0625)-1.0
              IF (T.EQ.0.) THEN
                    X(I)=1.0
              ELSE
                    X(I)=SIN(2.0*PI*0.5*T)/(PI*T)
              ENDIF
              WRITE (9,50) X(I)
20      CONTINUE
50      FORMAT (F20.5)
        CLOSE (UNIT=9)
        STOP
        END
```

Figure 3.31 FORTRAN Program to generate samples.

1. Set the alphabetic prefix for the data files to be used in this analysis and create and open a file to store data from an external file.

Input: FIL ANDQ

ILS responds with: Alphabetic characters set to: DQ

Input: FIL DEYDQ100

ILS responds with: TQ100. DOES NOT EXIST
PRIMARY FILE

Input: OPN

```
.00000
.06624
.13321
.21765
.30011
.38497
.47053
.55501
.63662
.71359
.78421
.84693
.90032
.94317
.97450
.99359
1.00000
.39359
.97450
.94317
.90032
.84693
.78421
.71359
.63562
.55501
.47053
.38497
.30011
.21765
.13921
.06624
```

Figure 3.32 Sampled Data from FORTRAN Program.

ILS responds with: (A system prompt.)

2. With the file DQ100 opened and ready to receive record data, the WRT command can be used to transfer data from the file WRTIN.DAT. The alphabetic argument used by the WRT command is T, which tells ILS to store real time series data from a file named WRTIN.DAT. The first numeric argument of the command identifies the number of items to store in each record, the second identifies the number of elements in each record, the third identifies the number of records to to use from WRTIN.DAT, and the fourth and fifth are the integer multiple of the sampling frequency of the record and the power of ten multiplier of the sampling frequency.

Input: WRT T32,1,32,16,0

ILS responds with: DQ100. RECORD 1 STORED

3. Using LRE, the data transfered to DQ100 is checked. The data is not from -1.0 to 1.0 as expected. (This is because the version of ILS used in performing this problem did not allow signals to be represented in the negative domain. Version 6.0, the newer version, does allow this.) This does not affect the solution of the problem.

Input: LRE

ILS responds with: Figure 3.33, a listing of data transfered from the external file, WRTIN.DAT, to ILS and stored as a record in file DQ100.

```
C:\ILS>LRE
******************************************************************
* C:\ILS\DQ100.                                 100,        1  RECORDS *
******************************************************************

------------------------------------------------------------------
  RECORD      1, SAMPLING FREQUENCY     16, TYPE 1111

REAL TIME SERIES OF SAMPLED DATA

  INDEX        TIME          REAL VALUE

     1      .0000E+00      .0000E+00
     2     6.2500E-02     6.6240E-02
     3     1.2500E-01     1.3321E-01
     4     1.8750E-01     2.1765E-01
     5     2.5000E-01     3.0011E-01
     6     3.1250E-01     3.8497E-01
     7     3.7500E-01     4.7053E-01
     8     4.3750E-01     5.5501E-01
     9     5.0000E-01     6.3662E-01
    10     5.6250E-01     7.1359E-01
    11     6.2500E-01     7.8421E-01
ENTER C TO CONTINUE, E TO EXIT, N FOR NEXT RECORD
->N
RECORD      2 NOT FOUND
C:\ILS>
```

Figure 3.33 ILS Record Data from WRTIN.DAT

4. DRE is used to display the sampled data values. Part (a) is figure 3.34 and part (b) is figure 3.35.

5. Now the DFT is computed using a FFT since the number of data points is a power of two and this is record data. Since no alphabetic arguments are used with the FFT command, the results of the FFT are stored as real and complex data. The FIL command must also be used to open a secondary file which the FFT command uses to store its results. LRE is used again to check the results.

Input: FIL SDEYDQ101

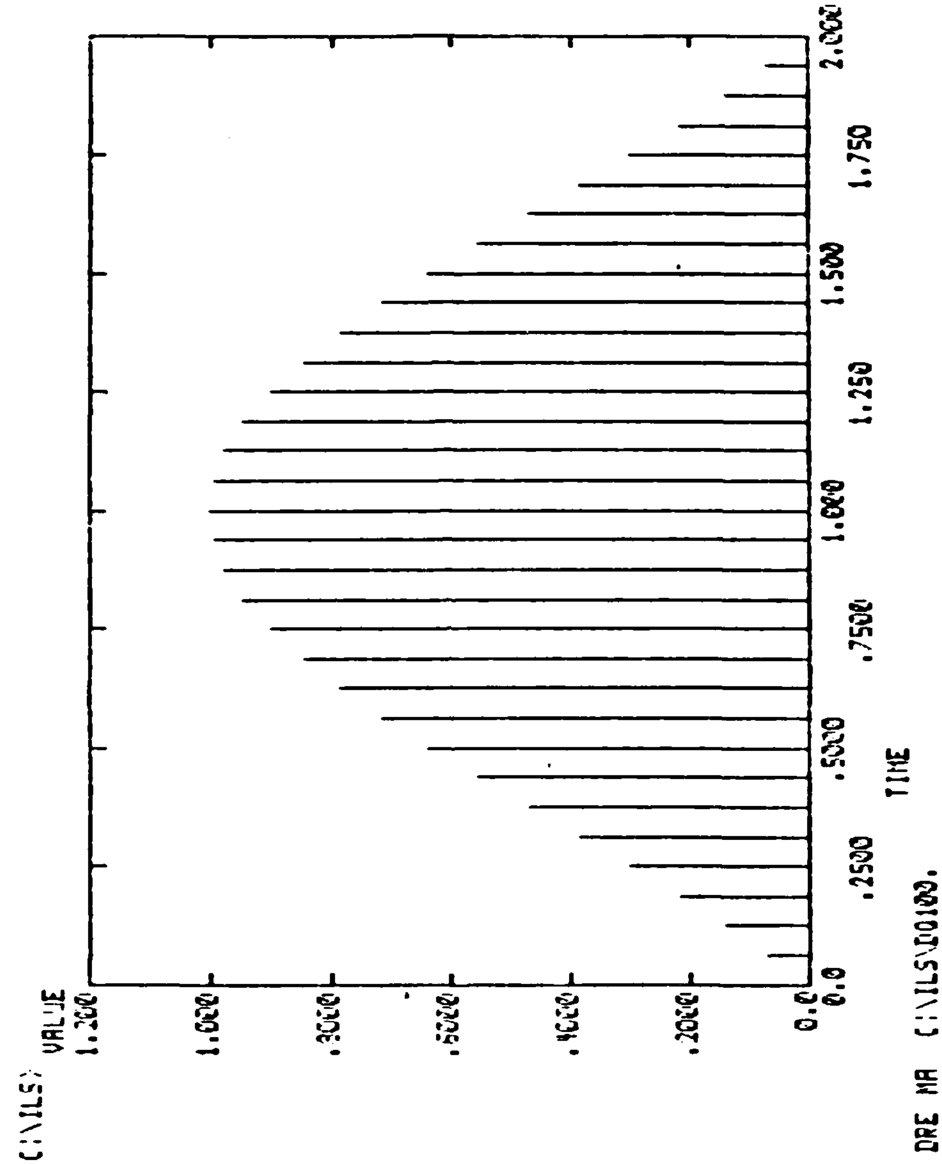

Figure 3.34 Input Data part (a).

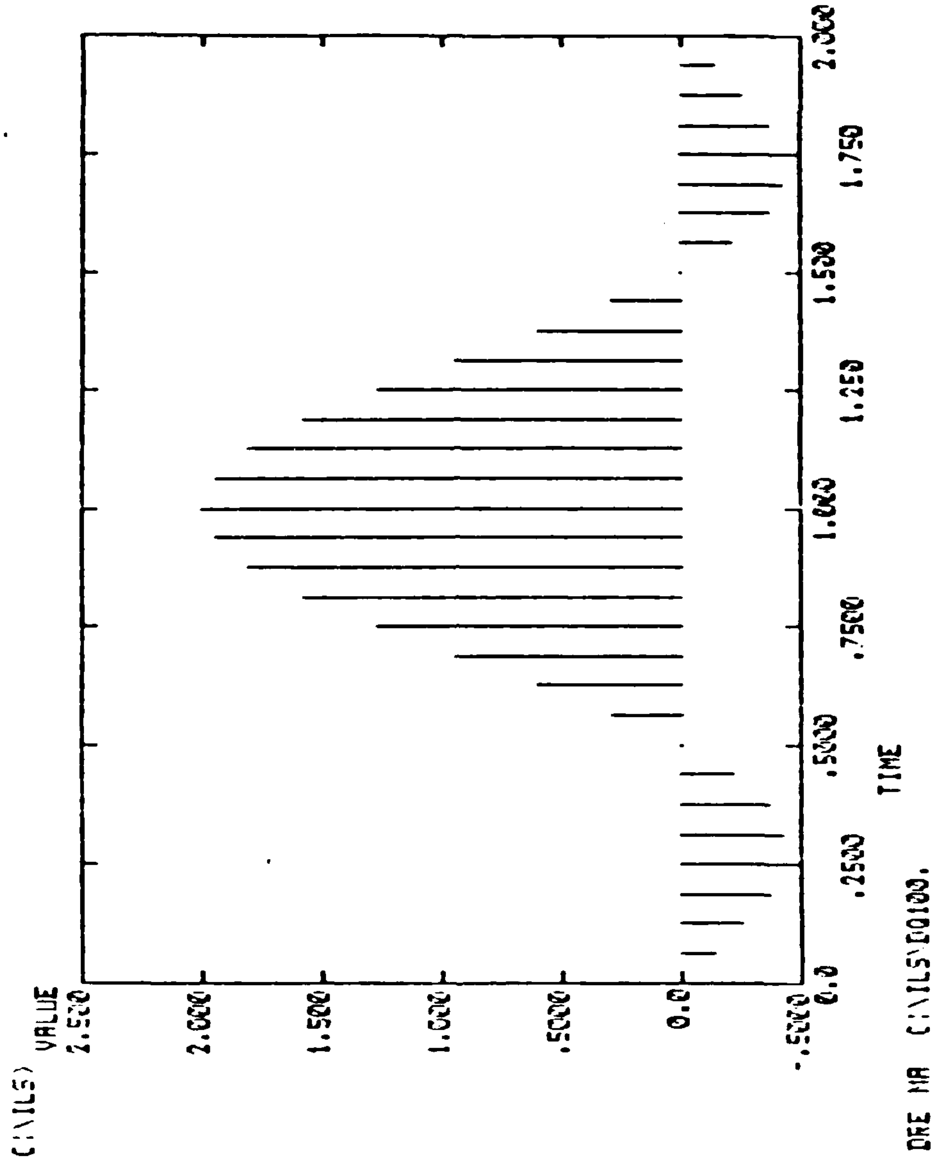

Figure 3.35 Input Data part (b).

ILS responds with: DQ101. DOES NOT EXIST
SECONDARY FILE

Input: OPN S

ILS responds with: (A system prompt.)

Input: FFT ,,5

ILS responds with: DQ101. RECORD 1 STORED

Input: LRE S

ILS responds with: Figure 3.36, the DFT of the sinc function.

```
C:\ILS>LRE S
*****************************************************************
* C:\ILS\DQ101.                                 101.       1 RECORDS *
*****************************************************************

-----------------------------------------------------------------
  RECORD       1, SAMPLING FREQUENCY      16, TYPE 2211

COMPLEX RECTANGULAR SYMMETRIC SPECTRUM OF SAMPLED DATA

  INDEX    FREQUENCY      REAL VALUE     IMAG VALUE

     1      .0000E+00     1.8853E+01      .0000E+00
     2     5.0000E-01    -7.2331E+00    -9.1446E-08
     3     1.0000E+00    -9.1285E-01    -1.9717E-08
     4     1.5000E+00    -3.8754E-01    -7.1470E-08
     5     2.0000E+00    -2.1854E-01      .0000E+00
     6     2.5000E+00    -1.4269E-01    -7.4784E-08
     7     3.0000E+00    -1.0224E-01    -2.6815E-08
     8     3.5000E+00    -7.8155E-02    -4.8791E-08
     9     4.0000E+00    -6.2800E-02      .0000E+00
    10     4.5000E+00    -5.2505E-02    -2.8721E-07
    11     5.0000E+00    -4.5346E-02    -1.1914E-08
ENTER C TO CONTINUE, E TO EXIT, N FOR NEXT RECORD
->N
RECORD      2 NOT FOUND
C:\ILS>
```

Figure 3.36 DFT Coefficients part (a).

6. The output file DQ101 contains the real and imaginary parts of the DFT but not the magnitude. This file can be modified to contain the magnitude of the DFT by using the UOP command. The UOP command does this by computing the absolute value of the data in DQ101 and storing the results in a secondary file. First the FIL command is used to change DQ101 to a primary file and initialize a secondary file. The OPN command opens the secondary file to accept the results of the UOP command. The UOP command uses the AB argument to identify that it will compute the absolute value of the data in DQ101. Default values for the numeric values are used.

Input: FIL DQ101

ILS responds with: DQ101.   RECORD DATA
13 DK BLKS.   1 RECORDS
PRIMARY FILE

Input: FIL SDEYDQ102

ILS responds with: DQ102.   DOES NOT EXIST
SECONDARY FILE

Input: OPN S

ILS responds with: (A system prompt.)

Input: UOP AB

ILS responds with: DQ102. RECORD 1 STORED

7. The DRE command is used to plot the magnitude of the DFT.

The second problem is performed in a similar fashion. Figures 3.37 and 3.38 are the plots of DFT of parts (a) and (b), respectively. [Ref. 4]

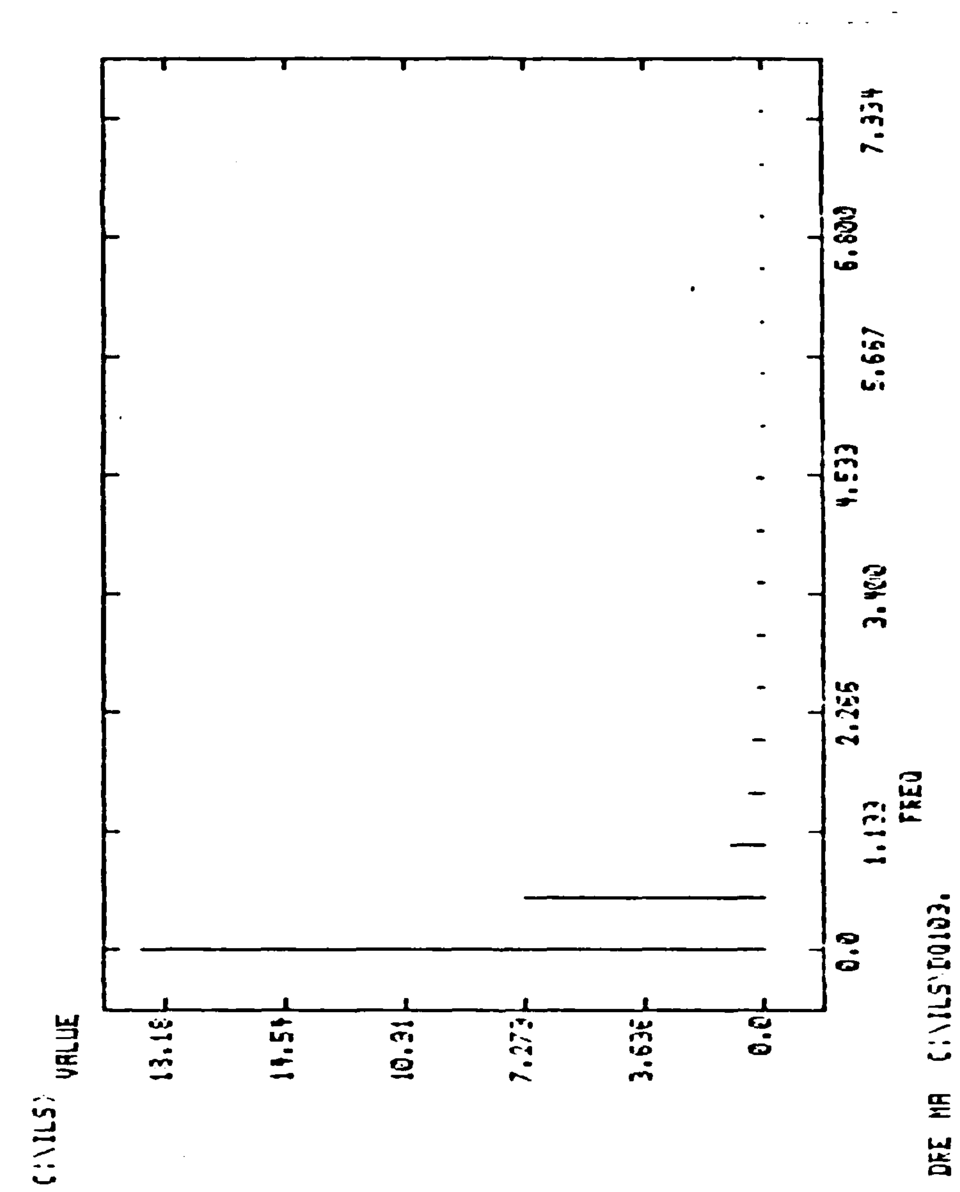

Figure 3.37 DFT Coefficients part (a).

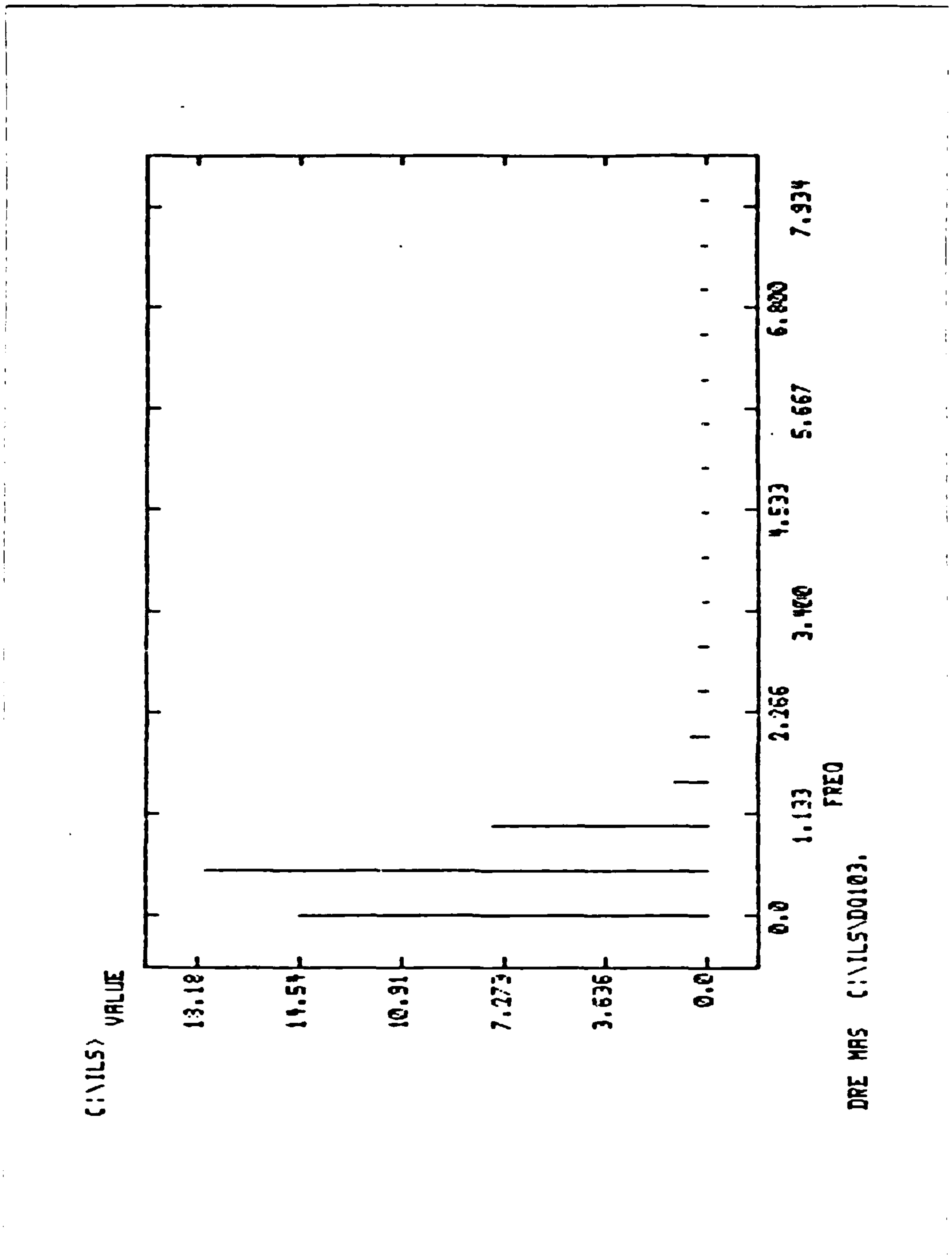

Figure 3.38 DFT Coefficients part (b).

IV. CONCLUSIONS

Having described the capabilities of ILS and worked several example problems, an evaluation of the system is appropriate. The software package is evaluated in an academic environment which consists of comparing the advantages and disadvantages of ILS to existing digital signal processing tools at the Naval Postgraduate School.

A. DIGITAL SIGNAL PROCESSING TOOLS AT NPS

Presently, only the mainframe system has programs which perform digital signal processing. These programs are available to the students by accessing a public disk. The primary signal processing applications available through these programs are:

- Fast Fourier Transforms.
- Discrete Fourier Transforms.
- Frequency response of continuous and discrete systems.
- Convolution.
- Correlation.
- Difference equation solution.

The programs are not interactive and require creation of output files and reprograming of subroutines within the programs in some cases. Plotting of the results from these programs is obtained using DISSPLA or EASYPLOT.

The mainframe is compatible with a number of programing languages such as FORTRAN, PASCAL, and APL. This allows the user to write signal processing programs for applications which are not available through the public disk. One of the more useful languages for application multidimensional signal processing analysis is APL. APL is used primarily for solutions which require an iterative matrix type of solution process. The results of an APL session are easily plotted using GRAFSTAT, an interactive plotting, data analysis, and applied statistics software package which uses APL.

B. ADVANTAGES AND DISADVANTAGES OF ILS

A comparison of PC-based ILS with existing mainframe signal processing tools identifies the advantages and disadvantages of both systems. First, ILS has the advantage of being readily accessible. There are no unscheduled or scheduled

maintenance requirements which affect the availability of ILS running on a personal computer. ILS is completely interactive so no substantial programing ability is required to use the software. The software can also be used in a menu driven environment. This allows the user to obtain a better understanding of the ILS commands and their usage in digital signal processing applications. The data generated through the use of ILS commands and stored in ILS files is easily plotted with ILS display commands. The ILS software can get data from external sources. The software has the ability to convert and use data from an external data file and with A/D and D/A boards, convert externally generated data to ILS format for processing and then reconvert and output that data in its original format. This gives the user an infinite variety of signals to generate and analyze.

There are some disadvantages with ILS, however. First ILS cannot perform multidimensional signal processing. The software limits the number of techniques available for performing signal processing analysis. For example, the user cannot design a recursive filter by other than the bilinear transform method. ILS does not integrate with other programs or software. Thus, it is not possible to incorporate new user-developed operations in the software. The graphics provided by the software does not display well with dot matrix type printers and its plots are limited to the software options and capabilities. For example, ILS will not smooth plots with curve fitting options such as an cubic spline, it merely connects the points. Finally, the user must have a very good understanding of digital signal processing techniques to apply the software correctly and interpret ILS results.

C. RECOMMENDATION FOR EMPLOYMENT

ILS is a powerful signal processing tool. The software requires the user to have a good foundation in signal processing techniques in order to use the system effectively. The menu driven and interactive capability of ILS makes this version of the software easier to use for beginners. Since the software consolidates a number of signal processing methods, it would be utilized most efficiently as a research tool. ILS could also perform as a lab tool for projects in courses which deal with spectral analysis, digital filter design, and speech processing.

APPENDIX
ILS COMMAND: DFI

This appendix consists of an excerpt from the ILS Command Reference Guide. It is representative of the documentation provided to the user concerning the capabilities, usage and format of ILS commands.

DFI

DESIGN FILTER COMMAND

Function

Design an elliptic, Butterworth, or Chebychev filter.

Command Format

```
DFI [P,S] [O] N1,N2,N3,N4,N5,N6,N7,N8
```

Input

The command arguments are the only input.

Output

An elliptic, Butterworth, or Chebychev filter matching the input specifications is stored in the COMMON file. The rational form filter can also be stored in a record file.

Alphabetic Arguments

O Design band pass filters for fractional octave-band filtering. (This argument is available only if the user's version of ILS includes the OCT command.)

Default is standard filter design (i.e. no fractional octave-band filters).

Select file in which to store the filter.

Default	stores the filter in the COMMON file only.
P	Store the filter as a record in a primary file also.
S	Store the filter as a record in a secondary file also.

Numeric Arguments

N1 The filter order (in S-plane).

Default is to prompt user for filter specifications.
Range is 1 to 14.

N2 The passband ripple in milli-dB.

Use >0 to indicate an elliptic or Chebychev I filter.
Use 0 to indicate a Butterworth or Chebychev II filter (neither of which has ripples in the passband).
Range is 1 to 10000.

N3 The integer sampling frequency in Hz.

No default.

N4 A band edge in Hz. For a low pass filter, N4 must be zero. For a band pass filter, N4 must be the lower band edge. For a band reject filter, N4 must be the upper band edge. For a high pass filter, N4 must be larger than 0 but less than the folding frequency (half the sampling frequency). The band edge represents the passband edge for the elliptic and Chebychev I filters, a half power point for Butterworth filters, and the stopband edge for a Chebychev II filter.

Range is 0 to half of the sampling frequency.

N5 The second band edge in Hz. For a low pass filter, N5 must lie between N4 and the folding frequency. For a band pass filter, N5 must be the upper band edge. For a band reject filter, N5 must be the lower band edge. For a high pass filter, N5 must be larger than or equal to the folding frequency.

Range is 0 to half of the sampling frequency.

N6 A stopband edge (in Hz) or "dB-down" (in dB).
- Use >0 to specify a stopband edge, which must lie outside of the passband.
- Use 0 to specify no ripples in the stopband (filter must be Butterworth or Chebychev I).
- Use <0 to specify a dB attenuation in the stopband.
- Range is 0 to one half of the sampling frequency when N6 is positive, and is -1 to -100 dB when N6 is negative.

N7 The power of 10 multiplier for frequencies.
- Range is -30 to +30.

N8 The number of bits of quantization.
- Range is 1 to 16.

If [O] is specified:

N4 The central frequency of the highest fractional octave-band.
- Range is 0 to half of the sampling frequency.

N5 The number of bands per octave.
- Default uses 1 band.
- Range is 1 to 12.

N6 The dB-down factor.
- Range is -1 to -100 dB.

N7 The power of 10 multiplier for frequencies.
- Range is -30 to +30.

Description:

The DFI command designs elliptic, Butterworth, or Chebychev (I or II) filters and stores them in the COMMON file. Each of these different types can be designed as a low pass, high pass, band pass, or band reject filter. Filter designs can be specified directly on the command line or interactively in response to prompts. The user can activate the interactive mode by entering DFI without any arguments.

The spectrum of the filter can be displayed with the FDI C command (refer to the example), and the filter can be used with commands such as FLT to filter sampled data or record data. If specified by the user, the filter is stored in the primary record file (P) or in the secondary record file (S) as well as in the COMMON file.

The filters are designed as close to the user's specifications as possible. These specifications include filter order, passband ripple, sampling frequency, passband edges, stopband edge and required attenuation in stopband. The filter coefficient design is printed at the end of the program (refer to the example). The numerator and denominator coefficients and the corresponding quadratic factors (with the zero order value of 1 suppressed) are stored in the COMMON file.

The printout consists of the following information:

1. Design specifications.

2. Denominator and numerator coefficients.

3. Poles and zeros (real and imaginary parts).

4. Quadratic factors for each pole or zero. Each complex pole or zero pair of the form [1 - (a+jb)z**-1], [1 - (a-jb)z**-1], where a and b are complex conjugate pole pairs, can be written as 1 - (2a)z**-1 + (a**2+b**2)z**-2. The first order quadratic factor is (-2a) and the second order quadratic factor is (a**2+b**2).

5. Time constant. The time constant is the number of samples for the impulse response of the pole closest to the unit circle to decay to 0.37 (1/e) of its initial value. In general, if the value exceeds 100, the filter may be ill-designed.

6. Noise bandwidth. The noise bandwidth is the theoretical equivalent bandwidth for white noise input. It is the value for which an ideal filter (gain of one in passband, zero otherwise) would produce the same energy.

If the filter order (N1) is not entered, the program prompts for the parameter values. The order of the filter is usually specified as less than 10 since higher order filters can produce numeric instabilities. An order of 3 or 4 is typical.

The type of filter designed, i.e., elliptic, Butterworth or Chebychev (I or II) is determined by the amount of passband ripple (N2) and stopband ripple (N6) allowed by the user.

	Passband ripple allowed, N2>0	Stopband ripple allowed, N6<0 or N6>0
elliptic	yes	yes
Butterworth	no	no
Chebychev	yes	no
Chebychev II	no	yes

For elliptic and Chebychev II filters (stopband ripple allowed), the user has a choice of specifying stopband edge in Hz (N6 >0) or the required attenuation in the stopband in dB (N6 <0).

The class of filter designed, i.e., low pass, high pass, band pass, or band reject, is determined by the values set for the lower passband limit (N4) and upper passband limit (N5). If these values are large, the power of 10 multiplier (N7) can be used.

Low Pass. To generate a low pass filter, the user specifies the lower passband limit as zero and the upper passband limit as greater than zero but less than the folding frequency, i.e., half the sampling frequency.

```
0-->|-------------|-------------|<--folding frequency (N3/2)
    |             |
    |             |
  N4=0        0<N5<N3/2
```

High Pass. To generate a high pass filter, the user specifies the lower passband limit as greater than zero but less than half the sampling frequency, while the upper passband limit is at least the folding frequency. If a frequency value greater than the folding frequency is entered as the upper passband limit, it is treated as the folding frequency.

```
0-->|-------------|-------------|<--folding frequency (N3/2)
                  |             |
                  |             |
          0<N4<N3/2     N5=N3/2 or N5_>N3/2
```

Band Pass To generate a band pass filter, the lower passband limit is specified as less than the upper passband limit while both fall between zero and half the sampling frequency. The user enters one value for N6, the stopband edge or required attenuation, the other value is determined by the program.

```
0-->|----------|----------|----------|<--folding frequency (N3/2)
               |          |
               |          |
        0<N4<N5   N4<N5<N3/2
```

Band Reject To generate a band reject filter, the upper passband limit is specified as less than the lower passband limit while both fall between zero and half the sampling frequency. The user enters one value for N6, the stopband edge or required attenuation, the other value is determined by the program.

```
0-->|----------|----------|----------|<--folding frequency (N3/2)
               |          |
               |          |
        0<N5<N4   N5<N4<N3/2
```

Octave band filtering

The 0 option is used to design filters for octave band filtering (see the OCT command description). The program designs one set of elliptic band pass filters and one elliptic low pass filter. The low pass filter has a cut-off frequency at the lower frequency edge of the highest octave band. The set of band pass filters are

designed with passbands adjacent to each other and with the total passband encompassing the entire band of the highest octave.

The numeric arguments specify the highest center frequency from the highest semi-octave band (N4), the number of elliptic band pass filters per octave band (N5), and the dB attenuation in the stopband (N6). The low pass filter is designed transparently. The order of the filter is one half of the band pass filters' order (i.e., N1/2) and the cutoff frequency is calculated as follows:

```
FC = [ 2 **(.5*M) ] * FH *.5

where:
   FC = high frequency cutoff of low pass filter
        (or, the lower frequency edge of the highest
        octave band)
   FH = the highest center frequency of band pass filters
        (or, N4 * 10** N7)
   M  = 1 / N5
```

General Notes:

1. Digital filters are designed from analog prototype low pass filters. For high pass and low pass filters the filter order is equal to the order of the digital filter. For band pass and band reject filters the digital filter order is twice the analog prototype filter order. A mapping is used to obtain the digital filter from the analog prototype filter. The mapping is based on the well-known bilinear transformation for low and high pass filters and on the lesser-known biquadratic transformation for band pass and band reject filters.

2. If quantization is selected (N8), a procedure is used to approximate a pseudofinite word-length filter. This procedure is as follows (please refer to the prior description of quadratic factors if necessary):

   a. the poles and zeroes are calculated in floating point notation (accuracy is system dependent);

   b. the pole/zero pairs are floating point multiplied to obtain quadratic factors;

   c. the first order quadratic factor is rounded within the range of +2 to -2 using the number of bits set by N8;

d. the second order quadratic factor is rounded with the range of 0 to +1 using the number of bits set by N8;

e. the poles and zeroes are recalculated in floating point notation from the rounded quadratic factors.

f. the numerator and denominator coefficients are recalculated using the above rounded quadratic factors.

Hence, both the cascade and rational forms of a filter generated by DFI approximate a pseudofinite word-length filter.

Example:

In this example, a low-pass filter with a cutoff frequency of 800 Hz and a sampling frequency of 10000 Hz is designed using two methods. First, a tenth-order Butterworth filter is designed, and a grid and the transfer function are plotted (Figure DFI-1). Second, a fourth-order Elliptic filter is designed, and the transfer function is plotted (Figure DFI-1). See OCT for an example of DFI 0 for octave band filters.

The design of the first filter is as follows. The order is set to ten and the passband ripple is set to zero. (No ripple means that the filter must be Butterworth or Chebychev II.) 10 KHz is the sampling frequency and the passband edges are set at 0 and 800 Hz. Since the stopband has zero ripple, the filter is a Butterworth low pass filter. The filter characteristics are printed after the DFI Command. NOTE: Each computer and operating system gives differing results on floating point calculations. The number of significant figures which agree varies widely, so the printed example may not match your own results. The differences show up most on the filter coefficients and least on the pole and zero locations.

```
DFI 10,,10000,0,800
DFI    10     0 10000     0    800     0     0

SAMPLING FREQUENCY        10000.000 HZ
LOW PASS BUTTERWORTH (MAXIMALLY FLAT) FILTER
BAND EDGES               800.000
```

	DENOMINATOR	NUMERATOR
1	1.000000E+00	2.440793E-07
2	-6.788961E+00	2.440793E-06
3	2.111861E+01	1.098357E-05
4	-3.954664E+01	2.928951E-05
5	4.928074E+01	5.125665E-05
6	-4.264121E+01	6.150798E-05
7	2.591558E+01	5.125665E-05
8	-1.091337E+01	2.928951E-05
9	3.045054E+00	1.098357E-05
10	-5.079836E-01	2.440793E-06
11	3.845145E-02	2.440793E-07

POLES		QUADRATIC FACTORS	
REAL	IMAGINARY	FIRST ORDER	SECOND ORDER
0.814894	0.442476	-1.629788	0.859837
0.719043	0.352213	-1.438087	0.641077
0.653642	0.254094	-1.307285	0.491812
0.613125	0.153026	-1.226251	0.399340
0.593775	0.051065	-1.187550	0.355177

ZEROS		QUADRATIC FACTORS	
REAL	IMAGINARY	FIRST ORDER	SECOND ORDER
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000
-1.000000	0.000000	1.000000	0.000000

TIME CONSTANT 13.244 SAMPLES
NOISE BANDWIDTH 800.335 HZ

FDI CE is used to display the filter response from the parameters stored in the COMMON file and erase the screen before the display (E) (Figure DFI-1).

```
FDI CE
```

A grid is placed with a border and a vertical dotted line at the 800 Hz point.

```
GRD 8,,800
```

DFI is used to design the second filter as a fourth-order filter, with a 200 mdB ripple in the passband (so that the filter must be either elliptic or Chebychev I), and a sampling frequency of 10 kHz. Since the band edges are entered in reverse order, N5 is less than N4, the filter is a stopband filter, with passband edges of 3000 Hz and 2000 Hz. N6=-50 specifies 50 dB attenuation in the stopband, so that the filter must be of a type with ripples in the stopband, making it an elliptic filter.

```
DFI 4,200,10000,3000,2000,-50
DFI        4    200 10000   3000   2000     -50       0

BND RJCT ELLIPTIC FILTER
PASS BAND RIPPLE                 0.200 DB
PASS BAND EDGES               2000.000      3000.000 HZ
STOP BAND EDGES               2276.582      2723.419 HZ
SAMPLING FREQUENCY           10000.000 HZ
ATTENUATION                    -50.000 DB

      DENOMINATOR      NUMERATOR
 1    1.000000E+00    4.489288E-01
 2    2.384186E-07    1.605494E-07
 3    2.392886E+00    1.759466E+00
 4    4.023314E-07    4.716140E-07
 5    2.429392E+00    2.621464E+00
 6    2.980232E-07    4.749588E-07
 7    1.146745E+00    1.759466E+00
```

```
 8    6.705523E-08    1.605494E-07
 9    2.331727E-01    4.489291E-01
               POLES                            QUADRATIC FACTORS
          REAL           IMAGINARY       FIRST ORDER      SECOND ORDER
       -0.268456         0.912634         0.536913          0.904969
        0.268456         0.912634        -0.536912          0.904969
       -0.221332         0.696132         0.442663          0.533587
        0.221332         0.696132        -0.442663          0.533587
               ZEROS                            QUADRATIC FACTORS
          REAL           IMAGINARY       FIRST ORDER      SECOND ORDER
       -0.130399         0.991462         0.260797          1.000000
        0.130399         0.991462        -0.260797          1.000000
       -0.056417         0.998407         0.112835          1.000000
        0.056417         0.998407        -0.112834          1.000000

TIME CONSTANT                20.029 SAMPLES
NOISE BANDWIDTH            4021.208 HZ
```

The spectrum of the filter is displayed from the user's COMMON file and placed on top of the previous figure (Figure DFI-1).

```
FDI C
```

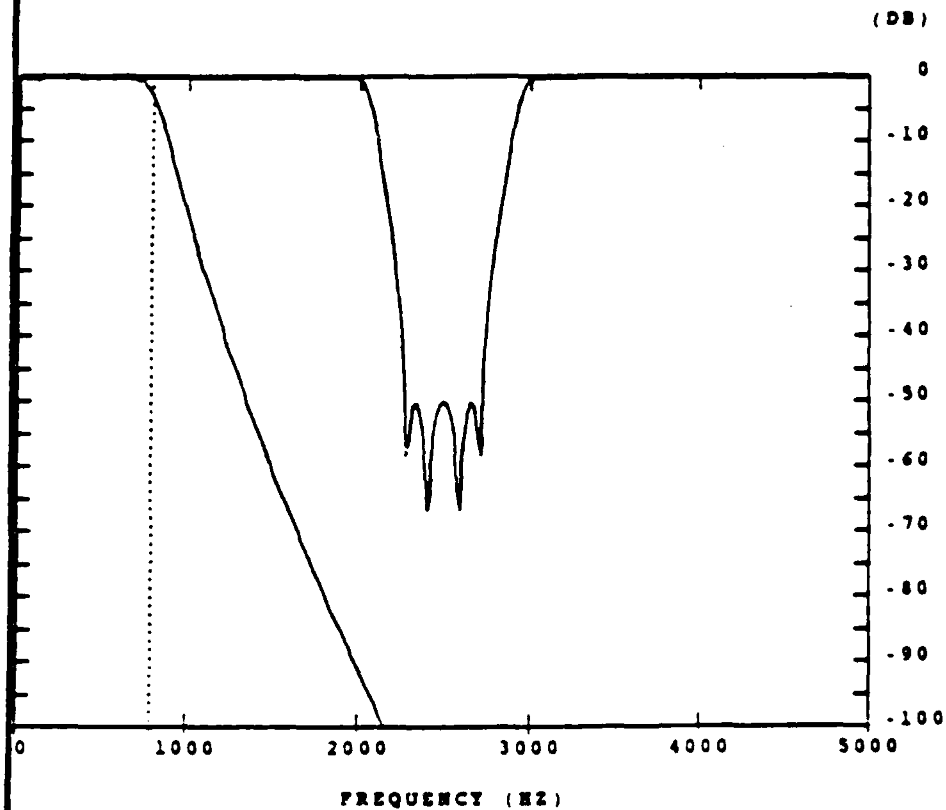

Figure DFI-1. Frequency spectra of the filters described in the example. The spectrum with the sidelobes is the elliptic design, the other is the Butterworth design.

LIST OF REFERENCES

1. Information in a letter to Professor Donald E. Kirk from Professor Charles W. Therrien Naval Postgraduate School. Monterey, CA dated 14 January 1987.

2. *ILS User's Guide V6.0 PC DOS*, Signal Technology Inc., Goleta, CA, 1987.

3. Strum, R. D. and Kirk, D. E., *First Principles of Discrete Systems and Digital Signal Processing* Addison-Wesley Publishing Company, Inc., Menlo Park, CA, 1988.

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